NEW! Version 1.5

The Big Book of HELIX

Tips & Tricks

All Helix Family Devices - Guitar & Bass - Free Presets & Files

Craig Anderton

“The Sweetwater Difference” and eBooks

Conventional print publishers are unable to cope with rapid changes in technology, nor is it fair to ask customers to buy books that are out of date almost the moment they’re published. Yet the need for accurate, authoritative, practical, and easy-to-understand information on gear is greater than ever.

Sweetwater has partnered with acclaimed author/musician Craig Anderton to re-invent publishing for technology-based books. The model is like software. The books are updated periodically, just like music software. New customers always get the latest version.

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Thank you for your support.

Chuck Surack
Founder and Chairman, Sweetwater

The Big Book of Helix Tips and Tricks
Version 1.5
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Preface

The Helix family of processors has been immensely successful. One reason is that the original floor Helix was deliberately overdesigned. Although this added to the price, over the years Line 6 has provided a steady flow of updates to expand Helix family devices with new models and features. Early adopters get new, free goodies. Those discovering Helix for the first time start with all the updates.

So why do we need a book about Helix?

Helix has options for creating your own signature sounds that have no parallel in other multi-effects. This eBook is loaded with programming techniques, as well as over 330 presets and files, to squeeze every ounce of performance out of Helix devices. You can also explore other Helix talents, like having it serve as a comprehensive computer audio interface. Perhaps most importantly, this eBook contains deep, detailed descriptions of the effects parameters—something Helix users have been wanting for years.

Like Sweetwater Publishing's other technology-oriented books, *The Big Book of Helix Tips and Tricks* receives periodic updates. In this case it's particularly relevant, because the latest firmware version (3.80) has further expanded the feature set. In addition to updating the book text, revising some presets from previous versions can take advantage of new features.

Furthermore, with increasing emphasis on immersive audio, many presets now incorporate stereo imaging and effects. This is ideal for the studio. However, most presets are also checked for mono compatibility so that they're suitable for stage setups.

I hope this book helps you discover the Helix family's vast potential, inspires your creativity, and helps you add a unique, personal twist to your music.

—Craig Anderton

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FAQ: About The Free eBook Updates

Why do you have an update program for eBooks?

Books about technology become outdated fast, and you need current information. To find out if an updated version is available, the following web pages list the latest eBook version numbers:

- The Sweetwater Publishing page
- The Products page on craiganderton.org, which lists the latest version numbers for all eBooks.

What's the difference between a "point" update and a new edition?

A "point" update (like software going from v1.0 to v1.1) is free. It adds new information, enhances the content, checks that links remain correct, fixes known typos, and when possible, incorporates suggestions from readers.

A new edition is like software going from version 1.x to version 2.0, and represents a significant change. New editions are not free, but are available at a reduced price for owners of previous editions.

How do I obtain an update?

To download a "point" update, use the same activation code you used to download the book when you purchased it. Simply download it again. What's available is always the current version. If you misplaced the code, it should be available in your Sweetwater Account under Account History.

How often are books updated?

There's no fixed schedule. Some titles haven't been updated yet, some were updated within a few months. It all depends on how fast technology changes. This is the book's fourth free update.

The Ground Rules: *Read Me!*

There are 5,687,359 Helix videos on YouTube. Everyone has opinions about what's good, what's right, what's wrong, etc. As far as I'm concerned, *every one of those videos is right!* Because the people are talking about what works for *them.* However, they're not *you.* Learn how to edit Helix, and then trust your ears.

IMPORTANT! The Potential, and Peril, of Presets

Presets that use distortion, or any preset that responds to dynamics (not only dynamics processors, but envelope filters and such) will sound different depending on the input level, the pickup type (humbucker vs. single-coil), which pickup is selected, your playing style, the musical genre, string gauge, etc.

This eBook's presets sound great with my guitars, and with input levels that tend to peak somewhat below where they would distort the input. So, if you're my clone with my taste in tones, the presets will sound wonderful. That's the potential of presets. The peril is that you're not my clone. If they don't sound wonderful to you:

- Check the input level going to the preset.
- Try using different pickups (e.g., neck instead of bridge or vice versa).
- If levels are too low to obtain the desired distortion or dynamics effects, increase the Drive for Distortion and Amp blocks, and lower the threshold for Dynamics blocks.
- If levels are too high, do the reverse—less Drive and a higher Threshold.
- Most Dynamics blocks in Helix Floor, Native, Rack, and LT include gain reduction meters. Exceptions are Autoswell and Feedbacker. Some preset descriptions give a recommended gain reduction. Adjust the input level or Dynamics block threshold to obtain this amount of gain reduction.
- A preset may not suit your tastes or musical style. Hopefully there are enough presets that you'll find ones that work well with your music. If not, after reading this book you'll know how to edit them to your exact liking.

Stomp Switch, Expression Pedal, and Snapshot Assignments

With a few exceptions for recommended expression pedal assignments that are important to the preset, you'll need to customize the presets for your specific workflow, desired snapshots, Helix family device, and setup. For example, with HX Stomp you might use two expression pedals, or one expression pedal and a footswitch.

Studio vs. Stage

Presets that sound great when recording may not sound the same on stage. It's often best to simplify studio presets for onstage use, like optimizing the stage version for mono.

Some of These Presets Are Not Like the Others

Much of this book is about obtaining sounds that are difficult to obtain with conventional guitar amps. I hope you'll like these kinds of sounds as much as the traditional sounds the eBook covers. In any case, we'll cover techniques you can adapt to creating all kinds of sound—from traditional to futuristic.

This Book Isn't "Helix 101"—That Material Already Exists

I've tried to avoid duplicating material that already exists, especially topics that are already covered well in the manuals. This book was written in the same spirit as my book *Electronic Projects for Musicians*—to give you the tools and tips to produce your own sounds. Over 330 presets and files help jump-start the learning process.

Expert vs. Beginner

This book has a mix of beginning, intermediate, and advanced material. Beginners can just load presets and ignore the tech info. Advanced readers can do deep dives. Hopefully, there's something for everyone. However, please let me know via one of the options described below if you have any ideas on how to improve this book. Book updates often include reader suggestions. It's great to be able to write a book that interacts with its audience. I read all your comments and try my best to respond.

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The book has multiple internal links. These jump to a subject for more information. With Adobe Acrobat Reader, a memory stack keeps track of multiple previous and next destinations. Type ⌘/**Alt+left arrow** to return to a previous destination, or ⌘/**Alt+right arrow** to return to a later destination. If the Windows version of Acrobat was modified to disable this navigation, do the following:

1. Right-click on the **Toolbar menu** at the top.
2. Hover over **Show Page Navigation Tools**.
3. On the Flyout, click on **Previous View** and **Next View** to enable these options. This adds their respective Left Arrow and Right Arrow icons to the Toolbar. ⌘/**Alt+left arrow** or ⌘/**Alt+right arrow** now function as described above.

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Kindles that use E Ink technology (e.g., Kindle Paperwhite, Kindle Oasis) display books in black and white. Kindle Fires are more like tablets and can display in full color.

Additional Notes

- Sections marked with **(1.5)** are new, or have been modified substantially, since eBook version 1.4.
- The book refers to HX Stomp and HX Effects as "2-Path" devices. Helix Floor, LT, Rack, and Native are "4-Path" devices.
- 2-Path-compatible presets are saved in that format but can open in 4-Path devices.
- Capitalizing Amp or Cab refers to specific Helix blocks. Lower-case amps or cabs are generic.
- Names in **Green Bold** are the names of files in the Free Files folder.
- A preset name with a dot (e.g., **3B.Jazz Rivet**) is compatible only with 4-Path devices. A preset name with an underscore (e.g., **CB_MandrinPreamp**), or no underscore or dot, is compatible with 2-Path and 4-Path devices.

Where can I give suggestions for future updates? What if I have questions, or there's something in the book I don't understand?

Contact me directly at craig.tech.center@gmail.com. Your feedback is what makes writing these books an interactive experience with *you*—the reader. Several updates in the books are a direct result of reader feedback and requests. I read *all* your comments and try my best to respond. Even if you just find a typo that needs correcting, or something that isn't clear, please let me know. *I want these books to be as helpful as possible.*

Chapter 1 | GENERAL TECHNIQUES

Before you start making sounds, you need to know how Helix handles processing power, the best ways to optimize levels, how to choose the correct input impedance, stereo vs. mono, and more.

Amp+Cab CPU Consumption

Digital hardware effects use digital signal processing (DSP) chips. These have a finite amount of processing power. Different multi-effects designs divide up this power differently.

With Helix's *dynamic* design, the effects draw whatever processing power they need. Some draw more than others, so you can decide which tradeoffs to make when creating a chain. If you choose effects that need lots of processing power, eventually you won't be able to add more effects. Substituting an effect that needs less power (like replacing a stereo processor with a mono version) reclaims some of that processing power.

Helix Floor, LT, and Rack ("4-Path" Devices)

The Floor, LT, Rack models have two main signal paths, Path 1 and Path 2. Each has its own dedicated signal processing chip. The two paths can split further into two *parallel* sub-paths (1A and 1B, 2A and 2B), for four parallel paths total. The sub-paths draw their power from the main path's DSP chip. The software-based Helix Native plug-in also has four paths. For more about parallel paths, see Input Split Block.

Each of the four paths has 8 blocks for inserting effects, for 32 total effects (fig. 1.1). However, because each chip has a finite amount of processing power, you likely can't create a preset with 32 complex effects.

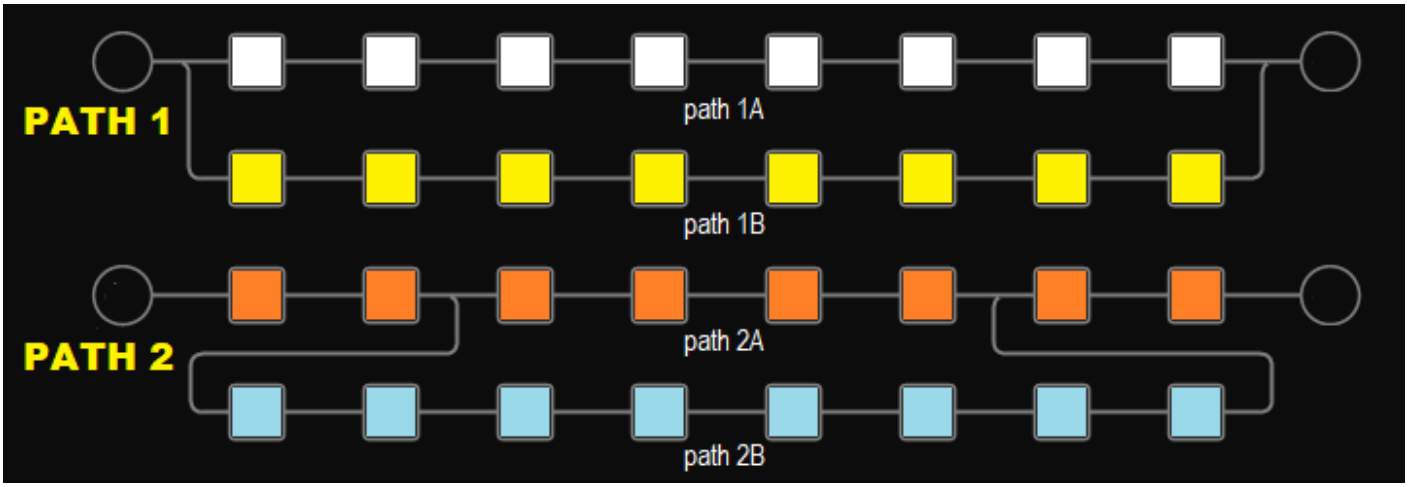


Figure 1.1 Path 1 can split into two parallel sub-paths, 1A (represented by white blocks) and 1B (yellow blocks). Path 2 can split into two parallel sub-paths, 2A (orange blocks) and 2B (blue blocks).

If you run out of processing power, consider these options:

- A 4-Path device can route the two main paths in series. This creates a single path that doubles the processing power of an individual path.
- Use a parallel routing. Distribute power-hungry effects between paths 1 and 2.
- Use some of the processing power-saving tips described in this eBook.
- When inserting effects, ones that can't be added are grayed-out. However, effects that aren't grayed-out may be able to do what you need. For example, if you need only a basic compressor, one that doesn't need much processing power can probably suffice.

The computer running Helix Native is the only limit on the plugin's processing power. So, Helix Native can create presets that won't run on Helix hardware. Fortunately, Helix Native has a preference that matches its processing power to specific Helix family devices. For more information, see Helix Family Compatibility.

HX Stomp and HX Effects

HX Stomp and HX Effects have a single DSP chip and one main path. This path can split into two parallel sub-paths. The two paths *together* accommodate a maximum of 8 blocks, for 8 effects total (fig. 1.2). The original HX Stomp allowed only 6 blocks. Although a software update upgraded this to 8 blocks, the DSP power remained the same. A preset that didn't fit in 6 blocks would not fit in 8. With effects that don't need much processing power, the two extra blocks allow for more sophisticated presets.

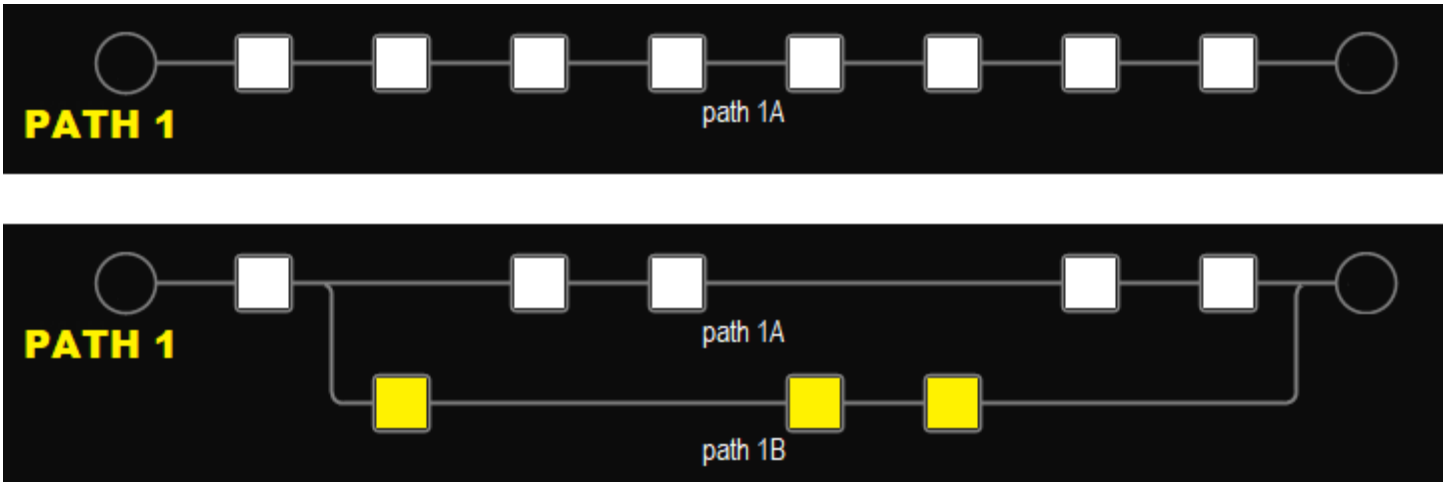


Figure 1.2 With HX Stomp/Effects, a single path (top) can hold 8 effects. It's also possible to create a parallel path (bottom, represented by yellow blocks). Unlike 4-Path devices, blocks in parallel paths aren't displayed directly above or below another block. They're always arranged linearly.

Here's an example of how Helix's dynamic design can trade off DSP power. Suppose you want an HX Stomp preset with the Wringer Fuzz, followed by the Revv Gen Red Amp+Cab and the Grammatico Nrm Amp+Cab in parallel. The Wringer Fuzz and Rev Gen Red Amp+Cab won't leave enough DSP power for the Grammatico Nrm Amp+Cab. However, using any of 33 other distortion processors frees up enough power to insert both Amp+Cabs in parallel. Only 8 DSP-hungry distortion devices won't be available. Or, save processing power by routing two amps in parallel. At the Mixer output, feed a single cab (see Share a Cab in Chapter 5).

Amp+Cab Blocks and Processing Power

The power each effect draws can change over different software revisions. Software coders like to optimize their code for efficiency, and Line 6's engineers are no different. Version 3.50 brought improvements to the Helix Core software that allowed new cab versions to draw around 66-80% *less* power than the earlier hybrid cabs. That alone is impressive, but they also sound more detailed.

However, not all legacy cabs are available as revised versions. Also, these improvements do not affect the Legacy cabs. Going forward, I recommend:

- Use the new cabs in new presets, to improve the sound quality and allow adding more processors.
- Substitute new cabs in presets with Legacy cabs, if the sounds are sufficiently similar. Dual cab blocks with the new cabs use less DSP than the Legacy single-cab blocks. So, you may be able to improve the overall preset sound or create stereo effects you couldn't do otherwise.
- Continue using Legacy cabs if new versions don't exist, or you like the Legacy cab sound better.
- Create Impulse Responses of the Legacy cabs to retain the sound and save DSP power.

Even IR blocks benefit from these changes. The Dual IR block introduced in version 3.50 draws less DSP power than a single IR block with the previous firmware. However, although IR blocks require less processing power, they still consume a lot of memory. So, they're still limited to one instance for each path.

You have many options for distributing processing power among a preset's effects. The following guide gives a (very) general idea of the DSP required by the guitar and bass Amp+Cab combinations. Don't obsess over this! If Helix runs out of processing power, you may be able to substitute a lower-power effect or amp, or use mono instead of stereo effects, and still have a preset that sounds like what you want. The following Amp+Cab groups are ranked from *lowest to highest* processing power consumption.

Group 1: Soup Pro, US Small Tweed, Jazz Rivet 120, Line 6 Electrik, Line 6 Litigator

Group 2: Tweed Blues Nrm/Brt, US Deluxe Nrm, US Double Nrm, Divided Duo, Interstate Zed, Essex A15/A30, A30 Farm Nrm/Brt, Matchstick Ch1/Ch2, Brit J45 Nrm/Brt, Brit 75 Nrm/Brt, Placater Clean, Archetype Clean, Line 6 Doom, Line 6 Epic, Line 6 2204 Mod, Line 6 Fatality, Line 6 Badonk, Line 6 Ventoux

Group 3: Stone Age 185, Voltage Queen, US Princess, US Deluxe Vib, US Double Vib, Mail Order Twin, Derailed Ingrid, Mandarin 80, Cartographer, Cali Texas Ch 1, Cali IV Rhythm 1, Cali IV Rhythm 2, Archetype Lead, ANGL Meteor, Solo Lead Clean, Solo Lead Crunch, PV Panama, Das Benzin Mega, US Super Nrm, US Super Vibe

Group 4: Who Watt, Matchstick Jump, Brit Trem Nrm/Brt, Brit Plexi Nrm/Brt, Brit 2204, German Ubersonic, Cali IV Lead, Cali Rectifier, Moo)))n Nrm (3.50 cab), Moo)))n Brt (3.50 cab), German Xtra Blue

Group 5: Fullerton Nrm/Brt/Jmp, Placater Dirty, German Mahadeva, Cali Texas Ch 2, Moo)))n Jump

Group 6: Grammatico Nrm/Brt/Jmp, Mandarin Rocker, Brit Trem Jump, Brit Plexi Jump, Solo Lead OD, Revv Gen Purple, Revv Gen Red, Das Benzin Lead, PV Crunch, PV Vitriol, EV Panama Blue, EV Panama Red, German Xtra Red

As another point of reference, the Mandarin 80 Guitar + Legacy is in Group 3. The 3.50 version, with the same amp but a revised version of the Legacy's cabinet, is more like Group 2. Similarly, the Brit Trem Nrm Guitar + Legacy is in Group 2. The 3.50 equivalent is Group 1. As a rule of thumb, using a version 3.50 cab instead of a Legacy one will bump the Amp+Cab down a group. For more about the new cabs and cabs in general, please see Chapter 6.

Bass Amp+Cab Blocks and Processing Power

Group 1: Del Sol 300, Busy One Ch1, Busy One Ch2
Group 2: Woody Blue, Cali Bass
Group 3: Ampeg SVT 4 Pro, Agua 51, G Cougar 800
Group 4: Ampeg B 15NF, Cali 400 Ch1, Cali 400 Ch2, Busy One Jump
Group 5: Ampeg SVT Nrm, Ampeg SVT Brt

As with the Guitar+Amp Blocks, the Bass+Amp Blocks with version 3.50 cabs are far more efficient.

Additional Processing Power Generalizations

- Effects with heavy distortion usually require more processing power than clean effects.
- Bass Amp+Cab models tend to need less power than guitar models.
- Jumped amps need more power than non-jumped equivalents.
- Amps generally consume more processing power than effects.
- Legacy effects often draw less power, as do most of the Line 6 custom Amps.
- Stereo effects need more processing power than their mono versions.
- EQs, Filters, Wahs, and the Volume/Pan block require the least amount of power.
- Distortion, and blocks that shift pitch often require higher-than-average DSP power.
- Some stereo modulation and reverb effects are power-hungry.
- For a comparison of all Helix block DSP allocations, see Helix DSP Allocations at BenVesco.com.

Choose the amp and cab you want first. Then, start adding effects. If you run out of power, try alternate effects. If the effect you want to add is essential, other amps may be suitable.

Helix Family Compatibility

Many people have more than one Helix family device—like Helix Native and HX Stomp, so you can take your favorite studio presets to gigs. Fortunately, it's easy to translate presets among multiple Helix family devices. But be aware of potential compatibility issues. For example, trying to load a 4-Path Helix preset with 12 effects into a 2-Path HX Stomp won't work, unless you modify the preset.

Helix Native can load any Helix family presets and create presets that are limited only by your computer's processing power. However, you can "throttle back" Helix Native to emulate other Helix devices. This ensures compatibility between Helix Native presets and Helix-family hardware devices (fig. 1.3).

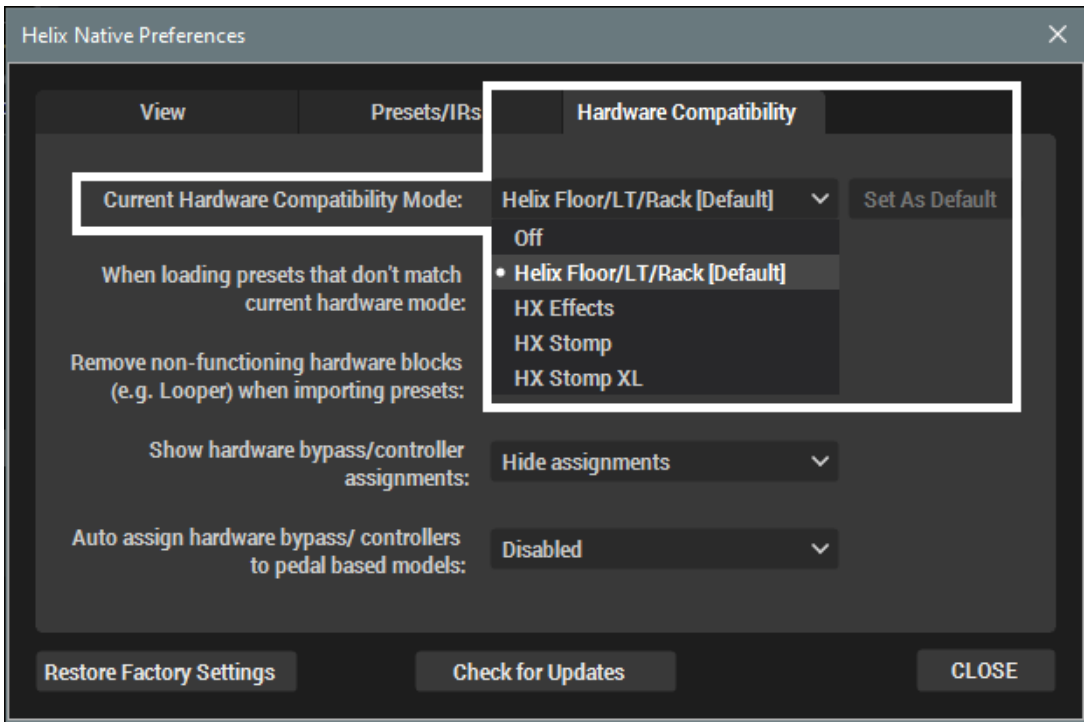


Figure 1.3 Helix Native's hardware compatibility mode can choose the same processing power constraints as various Helix-family hardware devices.

Helix Native's ability to emulate different Helix family devices makes it ideal for developing presets, because you can test them for compatibility with Helix family devices. HX Edit software:

- **Can** edit connected hardware devices.
- **Can** convert 2-Path Helix presets for use in 4-Path Helix devices.
- **Cannot** emulate different hardware.
- **Cannot** convert 4-Path Helix presets for use in 2-Path Helix devices.

About Compatibility Conflicts

Trying to load a preset from a hardware device that differs from Helix Native's current Hardware Compatibility mode may issue a warning (fig. 1.4). This can happen even with presets created on a Helix-family device with less processing power than the current mode. The dialog box offers two ways to retain compatibility:

- Change the current Hardware Compatibility mode to match the preset. This is best if you want the preset to be compatible with the hardware from which it came.
- Translate the preset to match the current Hardware Compatibility mode (if possible). For example, if the Compatibility mode is set to a 4-Path device, use this option if you want to use an HX Stomp preset in a 4-Path device.

Choosing Translate opens a warning about potentially modifying the preset. When loading a 2-Path preset into a 4-Path configuration, this warning doesn't matter because the preset you're loading won't use up the 4-Path device's DSP power. Effects may move to different blocks in a chain, but the effects' order won't change.

However, loading a 4-Path preset into a 2-Path configuration may modify some settings. Also, translation won't be possible if the preset requires more processing power, paths, or blocks than a 2-Path device provides.

When using Helix Native, I always select hardware compatibility for a specific hardware device. It's tempting to let Helix Native run wild, but you may create a preset that's usable only in a computer-based environment.

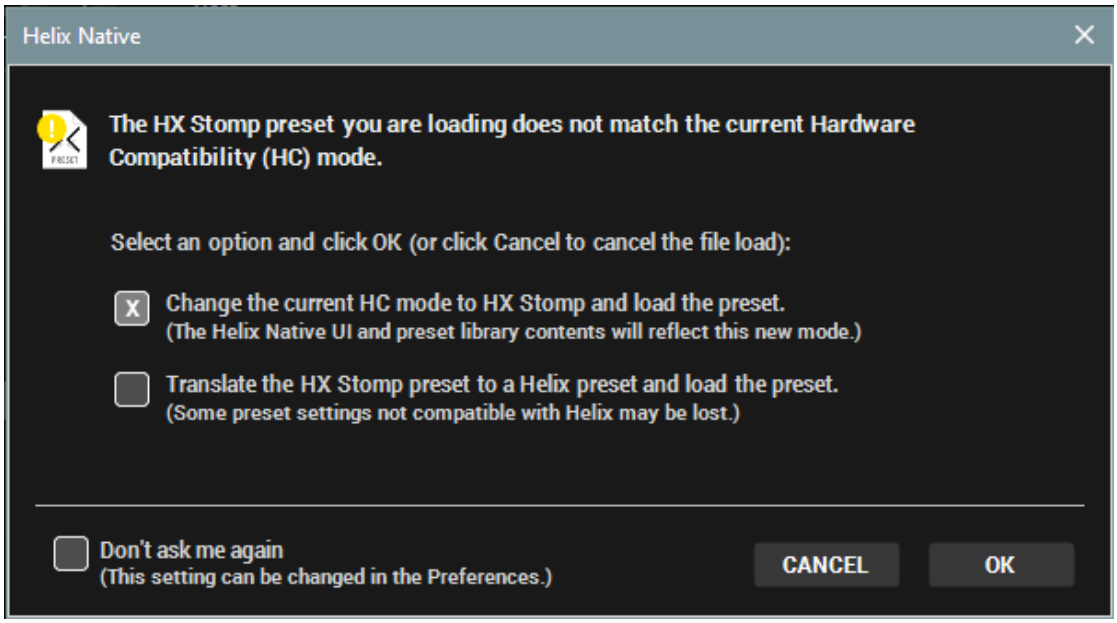

Helix Native
The HX Stomp preset you are loading does not match the current Hardware Compatibility (HC) mode.
Select an option and click OK (or click Cancel to cancel the file load):
X
Change the current HC mode to HX Stomp and load the preset.
(The Helix Native UI and preset library contents will reflect this new mode.)
Translate the HX Stomp preset to a Helix preset and load the preset.
(Some preset settings not compatible with Helix may be lost.)
Don't ask me again
(This setting can be changed in the Preferences.)
CANCEL
OK

Figure 1.4 This type of warning screen appears when you try to load a preset into Helix Native that doesn't match the current hardware compatibility mode.

Translate Incompatible Helix 4-Path Sounds to 2-Path Devices

You might develop a Helix 4-Path sound for use in the studio, but then want to translate it to, for example, an HX Stomp that you take to gigs as a backup. Try these options first:

- With HX Edit open to HX Stomp (or another 2-Path device), try opening a 4-Path preset. It may work. If it's not possible, you'll see a warning that "HX Edit was unable to import presets."
- With Helix Native, when set to HX Stomp hardware compatibility, try opening the preset. When the window appears that says the preset you're loading doesn't match the current hardware compatibility mode, check "Translate the Helix Native Preset to a HX Stomp preset and load the preset." This may work. However, an accurate translation may not be possible.

If the above methods don't work, you'll need to copy elements from the 4-Path preset over to the 2-Path device manually. Then you can reconstruct the preset as best you can. There are two ways to do this:

- If you have a hardware 4-Path and 2-Path device, both can be open at the same time in HX Edit. Drag-and-drop the most important blocks from the 4-Path preset to the 2-Path preset, until you run out of processing power. This preserves the block's settings.
- With Helix Native, open the 4-Path preset. Open the 2-Path device in HX Edit. Copy blocks from the 4-Path preset and paste them into a 2-Path preset in HX Edit. Note that you can't drag-and-drop from Helix Native to HX Edit. Only copying and pasting is possible.

Stereo and Mono

Helix family members are all true stereo in and out. Although guitars have mono outputs, Helix can generate a stereo output from a mono input. This includes vocals. Helix is also a great processor for keyboards, drum machines, and other stereo sound sources.

Mono/Stereo Audio Compatibility

If you create stereo presets for the studio, then expect to use these presets on stage, always check the preset's output when summed to mono. There could be phase cancellations that won't be noticed in stereo but may create a thinner sound in mono.

Tip: Checking for mono compatibility is always good practice, even when creating presets for the studio. A playback system (e.g., radio, pre-show music on a PA system) might be mono.

Live Performance

For live performance, a mono output makes sense—audience members toward the left side of the stage should hear the same music as those toward the right side. However, there are two exceptions where stereo presets may be preferable:

- A two-amp setup where the amps are close to each other. This isn't about stereo's spatial separation, but creating an overall bigger sound that mixes acoustically in the space where you're playing. It's still good to check for mono compatibility. Even though this is less crucial in a stage environment, your audience still hears what's essentially a mono signal.
- Streaming concerts or events. Home playback systems are usually stereo.

In the studio, stereo is usually preferable. However, not all effects have to be stereo. For example, Single cabs are mono. So, any effect *before* a Single cab can be mono. Then, adding a stereo effect after the cab creates a stereo image (or, use a Dual Cab). Following a stereo effect with stereo effects preserves the stereo image.

About Preset Levels, the Pad, and the Volume Control

Plugging into a digital device requires making some choices, so it's a different experience from just plugging into a physical guitar amp and turning up the volume.

- Helix has no physical input level control because guitar output levels are quite similar. The level-reducing pad switch can handle guitars with hotter outputs.
- Generally, leave the pad *off* for standard, passive guitar pickups. For extra-hot pickups or line-level signals, turn the pad *on* to drop the input level by about 6 dB. However, turning on the pad with passive pickups can also act like a global "turn down the drive" control. Some players using passive guitar pickups find Helix more touch-sensitive with the pad on. (Another option is to turn down an effect's drive parameter to make individual distortion-oriented blocks more touch-sensitive.) To compensate for a decreased output level due to the lower input level, you may need to increase a preset's output level.
- The main volume control is an attenuator, so it doesn't add gain. Its main use is lowering the output level, if needed. My preferred default setting is full up. This also gives the best fidelity when the volume control sets the level going to digital outputs.

The Input Impedance "Trap"

A device's input impedance interacts with passive guitar pickups, and alters their frequency response and level:

- Higher impedances preserve the pickup's treble response. Most audio interfaces have a high-impedance instrument input, so the sound entering your computer has the highest possible fidelity. Some guitarists prefer the clarity of a high-impedance input. Those who prefer a duller, less bright sound can add a high-cut filter before an effect.
- Lower impedances dull the highs. Some fuzz effects have a deliberately low input impedance to give a smoother, less harsh distortion effect.
- Impedance interacts only with the guitar itself. For example, if the first effect a guitar plugs into has a high impedance, and the next effect has a low impedance, the low impedance won't affect the guitar.
- Impedance matters less with line-level signals. A typical input impedance for these signals is 10k to 70k.

Helix takes a unique approach to accommodate those who like high impedances, and those who prefer the "native" sound of low-impedance effects. The Input icon for Guitar and Multi has an input impedance parameter that chooses among 8 input impedances (10k, 22k, 32k, 70k, 90k, 136k, 230k, 1 Meg). If your guitar plugs *directly* into Helix, your guitar will always "see" the chosen setting's impedance.

The Auto Option

In addition to a fixed impedance, an Auto Impedance mode causes amps and effects to emulate the input impedance of their physical counterparts. Auto Impedance mode has two options under Global Preferences:

- With *First Block*, your guitar sees the first effect's modeled impedance. For example, if the first effect is a fuzz with a 10k input impedance, the sound will be the same as if you'd chosen 10k as the input impedance. If the first effect has a 230k input impedance, the sound is the same as if you'd chosen 230k as the input impedance.

- A fine point with First Block is that Line 6 assumes effects do not use true bypass. So, if the first effect is 10k, *even when bypassed* your guitar sounds like it's feeding a 10k input impedance. You may not always want the dulling a fuzz produces, especially if you're feeding an effect where you want a brighter sound. In that case, choose First Enabled (see next).
- With *First Enabled,* the input impedance is the chain's first *enabled* effect. For example, suppose the first effect has a 1M input impedance and the second effect has a 10k input impedance. With this option, if the first effect is enabled, your guitar sees a 1M impedance. If it's bypassed but the second effect is enabled, your guitar sees the 10k input impedance.

Tech Talk: Linking Input Impedance to Particular Effects with Helix Floor

I prefer a high-impedance input, so I control the input impedance with the same footswitch that bypasses or enables an effect:

1. With the Guitar or Multi input selected, press Knob 4 (Guitar In-Z).
2. For the Parameter, choose Guitar In-Z.
3. For the Controller, choose the footswitch you've assigned to the effect.
4. The Min value will be the input impedance when bypassed (I set it to 1M), and the Max value, the input impedance with the effect enabled. I almost always set this to Auto.

Another option is to link the Guitar In-Z parameter to Snapshots. Then a Snapshot's impedance value can complement your effects' bypass/enable states.

The "trap" referred to earlier is that users may call up a Helix model of a device with a low impedance, but it doesn't sound the same as the hardware. To solve this, and have the effect sound as expected:

- Activate Auto's First Block mode and make this effect the first one in the chain.
- Or, activate First Enabled but don't precede it with a high impedance effect.
- Or, alter the input impedance to match the effect.

How to Level Presets

People have different attitudes about how (or even whether) to set presets to the same output level. There are four complications when trying to set consistent levels:

- A sound's *perceived* level can be different from its *measured* level. A bright, upper midrange sound has more energy where our ears are most sensitive compared to a bass sound. So, the upper midrange sound may *measure* as softer than a bass sound, but you may *perceive* the bright sound as louder.
- You *want* some sounds, like solos, to have a louder perceived level than others.
- With live performance, different presets have different perceived levels depending on room acoustics, the size of the audience, and the music you're playing.
- Presets that sound good in a home studio over monitors may not work well for live performance.

When using presets onstage, the only way to *guarantee* setting the right levels is to adjust them while playing live, in context. Regardless, having a consistent, baseline level speeds up the tweaking process.

Here's an analogy. When adjusting a pickup's pole pieces, I screw them all in halfway. Then if a string needs to be louder or softer, I can adjust the pole pieces as needed. If they'd all been screwed out, I couldn't make them louder. If they'd been screwed all the way in, I couldn't make them softer. It's easier to tweak preset levels if they're already close to what you want.

The following is intended for those who are familiar with recording, editing, or mastering audio. If your head explodes, move on to the About Gain-Staging section in this chapter.

A Partial Solution, Borrowed from Mastering Engineers

A measurement protocol called LUFS (Loudness Unit Full Scale) measures *perceived* loudness, not *absolute* loudness. The origin story (every superhero has an origin story, right?) is that the European Broadcast Union (EBU) had enough of mastering engineers making CDs as loud as possible to win the "loudness wars." LUFS measurements allow streaming services like YouTube, Spotify, Apple Music, and others to adjust the volume of various songs to the same *perceived* level. So, you don't have to change the volume for every song in a playlist—a Belgian hardcore techno cut from 1998 sounds like it's the same level as Billie Eilish. The system isn't perfect, but it's better than dealing with constant level variations.

LUFS meters measure perceived levels. Some DAWs include LUFS meters. Several third-party plug-in meters are available, like the Waves WLM Plus Loudness Meter, or the Youlean Loudness Meter (the company offers a free basic version).

The goal with leveling presets is for their outputs to have the same LUFS reading. This is *not* a panacea! You will almost certainly need to tweak output levels for specific performance situations and musical material. Having a standard output level simply makes tweaking easier, because you've established a standard. Presets may need to be either louder or softer than the standard level.

Setting the Output Level

If you don't care about whether the preset has the same perceived level as a dry guitar (you don't need to), adjust the output to whatever sounds right. However, I find comparing the processed sound to the bypassed sound is a useful baseline.

Creating consistent preset levels is *much* easier to do with a computer and Helix Native. Import the presets into Helix Native, tweak them, then transfer them back to your stage devices. Here's the process:

1. Set the Helix Native input and output levels to 0.0, *and don't touch them.* You'll make any needed input or output level adjustments in the preset itself.
2. Record a 15-30 second or so clip of bypassed guitar playing chords, without any major pauses, and another clip with 15-30 seconds of single notes, also without major pauses. For bass presets, record some bass lines.
3. Insert an LUFS meter after Helix Native. Depending on the preset, loop the chord or single note clip at least two or three times with Helix bypassed. Check the LUFS reading.
4. After enabling Helix, reset the LUFS meter, and again play the same loop at least two or three times. Adjust levels within the preset to hit the same target LUFS reading.
5. Whenever you change the preset level, *reset the LUFS meter reading* before playing back audio into Helix.
6. Use your ears to do any final output level edits, based on the musical context.

Tip: Most LUFS meters measure the *instantaneous* LUFS level as well as an *average* level over time. If you play the loop through a few times, the average level will settle to a final value, whether bypassed or through an effect. Use this reading, not the instantaneous one.

Note that this isn't an exact science. The object is for your presets to have a standard, baseline level. That way, when you get to the gig, massive tweaking probably won't be needed.

About Gain-Staging

Gain staging is the process of adjusting the gain through each effect in a chain for a consistent level. The object is to avoid having one effect overload another and cause unintended distortion (compared to the *intended* distortion we want from amps and effects). Another goal is to prevent input signals from having too soft a level, which could cause level-sensitive effects (like compressors) not to operate as expected.

With a *unity gain* strategy for gain-staging, an effect's perceived output level is the same as its input level. The advantage of this approach is that you can enable or bypass any number of effects, and the perceived level will be similar.

However, gain-staging can also mean choosing to have some effects at a softer or louder level when enabled, depending on the musical context (e.g., playing a solo). Just remember that the primary goal is to prevent overloading within an effects chain or when feeding external equipment.

Favorites, and Preset-to-Preset Block Copies

Saving a gain-staged block as a Favorite (fig. 1.5) lets you load it instantly—no tweaking required. The *Free Files >Chapter 11 Favorites* folder includes blocks I use a lot. They're described in Chapter 11.

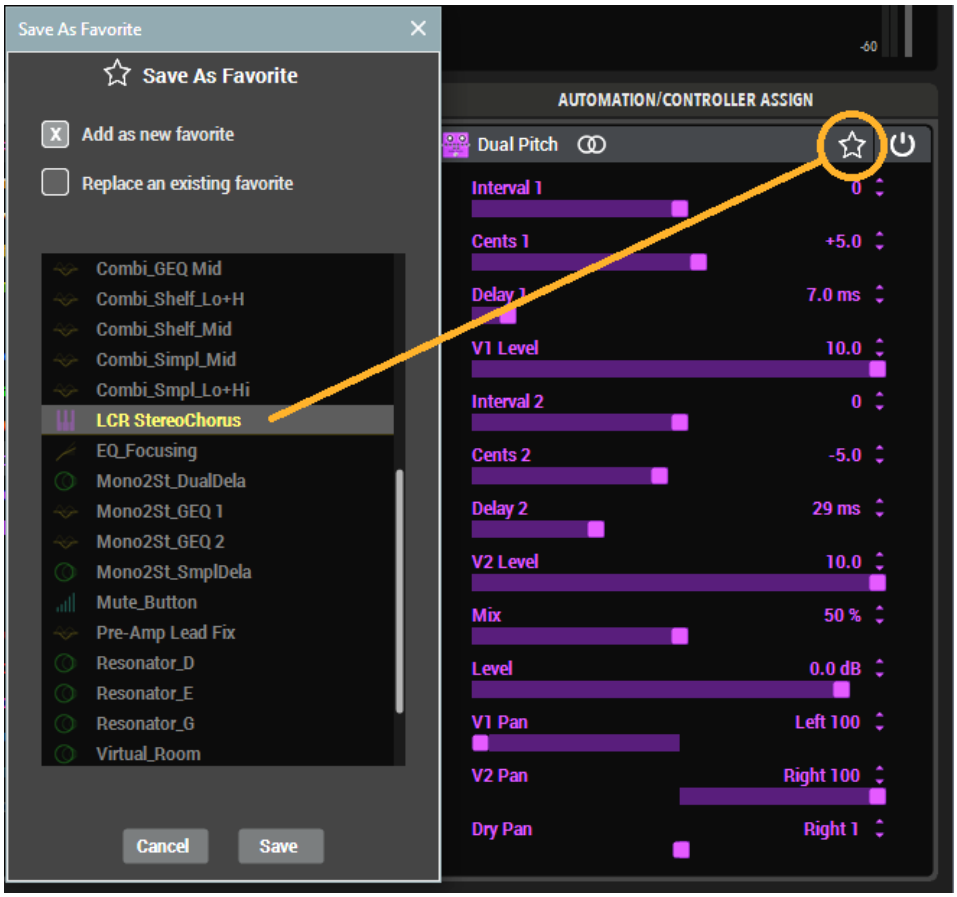


Figure 1.5 Click on a block's Star to save the settings in a Favorites menu. You can save, rename, clear, and replace up to 128 Favorites.

If a block isn't saved as a Favorite, but exists in a different preset, you can copy a block in one preset to a different preset. To copy and paste blocks between presets:

1. Save the preset you're creating.
2. Open the preset with the block you want to use.
3. Copy the block, then close the preset.
4. Re-open the preset you were creating. Paste the block in the desired path position.

You can simplify the process if you have more than one Helix-family device and HX Edit, or Helix Native:

1. Load the preset into a compatible Helix-family device in HX Edit (or Helix Native).
2. Open the preset. Copy the desired block.
3. Open another HX Edit window for the target device, or another instance of Helix Native.
4. Open the target preset in the HX Edit window or Helix Native.
5. Paste the block into the desired target preset block.

Mute One Path in a Preset with Parallel Paths

When developing presets with parallel paths, muting a path can help you concentrate on editing the other path. This can also help with creating consistent levels, by making it easier to edit the balance among paths.

Option 1: Mixer Level Controls

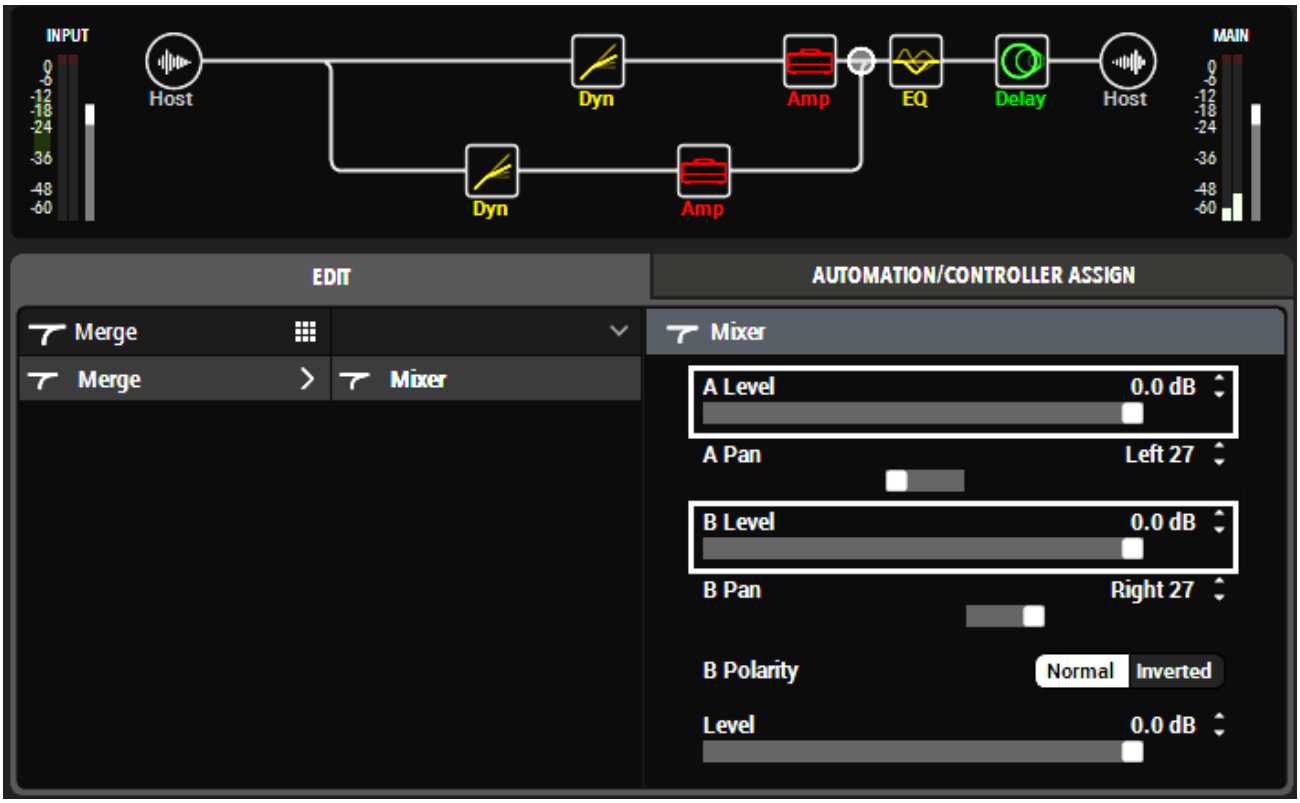


Figure 1.6 The A Level and B Level sliders control the levels of their respective paths.

Fig. 1.6 shows a preset that uses bi-amping techniques. To tweak the paths while creating a preset, you can turn down one path in the Mixer when you work on the other one. This approach has three main limitations:

- It can be inconvenient to keep opening and closing the mixer.
- If you've turned down a path with the Level control, you need to return it to the original, setting after turning down the path.
- With a 4-Path device preset that uses three or four parallel paths, you must open two different Mixers to do the editing.

Option 2: Create a "Mute Button"

A Volume/Pan block can act like a mute button. Although this uses up a block, after making the edits, you can remove the mute button blocks to insert other processors. The *Free Files > Chapter 11 Favorites* folder includes a Mute Button block.

To convert a Volume/Pan block into a mute button, set the Gain parameter to minimum (fig. 1.7). Enabling the Gain block mutes the audio because the volume is the minimum possible. Bypass the block to unmute.

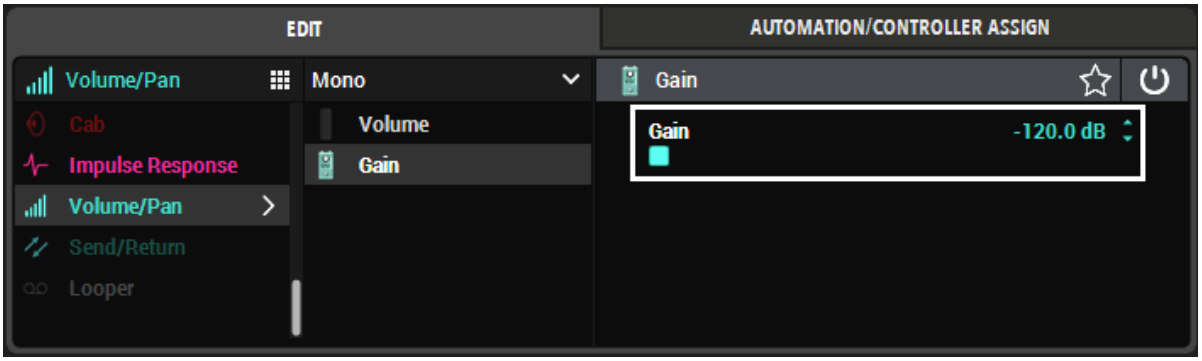


Figure 1.7 Setting Gain to -120.0 dB, and enabling the Volume/Pan block, mutes any audio going through it.

Fig. 1.8 shows a practical example of using mute buttons. This is a template for creating multiband presets with four bands.

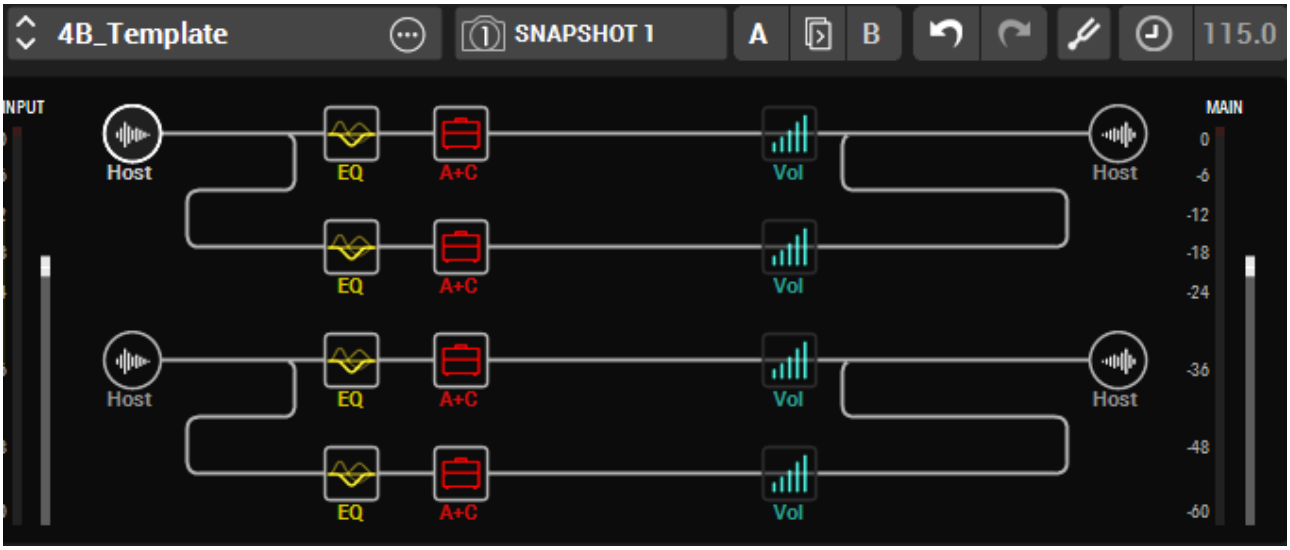


Figure 1.8 Multiband template, with a "mute button" for each path.

After splitting the input into frequency bands, you can optimize each Amp+Cab's sound for its associated frequency band. The Volume/Pan blocks make it easy to mute some bands, so you can audition other bands.

Chapter 2 | DE-MYSTIFYING THE BLOCKS

Although many block parameters are easy to figure out, some blocks—especially those in the Line 6 Originals—have unique, innovative parameters. Knowing how they affect the sound helps you get the most out of Helix.

Distortion Blocks

Which Is the Best Distortion?

Online forums often have debates about Helix's distortions. Some people haven't found their ideal distortion yet, and seek opinions of which distortion is "best." But one person's go-to distortion might be another's avoid-at-all-costs distortion because of:

- **Priorities.** A player may want distortion that's touch-sensitive over a wide range, so the guitar growls if you don't pick too hard but screams if you do. Others might want heavy distortion without much picking, so there's a consistent level. Do you want a swampy, bluesy tone? Something that cuts through a mix, or lays back in the mix? Vintage, or modern? Pop, hip-hop, British invasion, industrial, or EDM?
- **Strings, pick, and pickups.** These affect how distortion responds. Wound strings over steel cores give a vintage, fatter tone, while the more common hex cores sound brighter. Nickel windings sound warmer than harder metals, like chrome and steel. Pickups make a huge difference. Simply rolling down a tone control might tame an otherwise nasty fuzz. Even your pick, neck material, and frets matter.
- **Input level to the distortion.** Some distortions give more touch-sensitive, nuanced sounds when you reduce the drive, or roll back your guitar's volume control. Conversely, some distortions resist cleaning up—heavy distortion is in their DNA.
- **What follows the distortion.** Distortion responds differently when followed by different amps and cabs. Some amps, like the WhoWatt 100, seem more compatible with different distortions.
- **Personal taste and musical style.** I'm picky about distortion sounds, which is why I got into multiband distortion. But that's not for everyone—it's not even for *me* all the time. Sometimes I just like a single Amp+Cab, put distortion in front of it, and let 'er rip.

Diodes: Germanium vs. Silicon vs. LED

Many guitarists are confused about the difference between germanium and silicon diodes. You can generally assume any distortion uses silicon diodes unless stated otherwise. The first use of germanium diodes was in radar during WWII. Silicon diodes have *many* advantages over germanium diodes. Currently, they're used only in niche applications, mostly involving radio frequencies. For more information, see the Tech Talk on diode characteristics. There's also a discussion of LED vs. silicon-based fuzz in the Pillars OD description.

Also note that diodes of the same type can give different distortion characteristics, depending on how they're configured. See the Vita Dist description.

The only significant difference between germanium and silicon diodes is that germanium diodes clip at a lower level (0.3V for germanium, 0.7V for silicon). There's no significant difference in *tone* between the two types of diodes themselves, except for how those level changes may affect elements of the design that *aren't* diodes. Furthermore, virtual distortion is not like hardware because clipping isn't "baked into" the hardware. With virtual distortion, if you want something to clip at a lower level, just put more gain in front of it.

The Bottom Line

Play through every distortion. Go beyond the default preset by experimenting with the controls (and remember that sometimes, parameters interact). Learn what the controls do, and when you find an awesome combination of settings, save it as a Favorite.

Just remember: *Each distortion has different strengths and limitations.* Find out what they are, so you can pick the right fuzz for the right context. Also make sure you understand how impedance affects the distortion sound.

About the Basic Distortion Blocks Reference

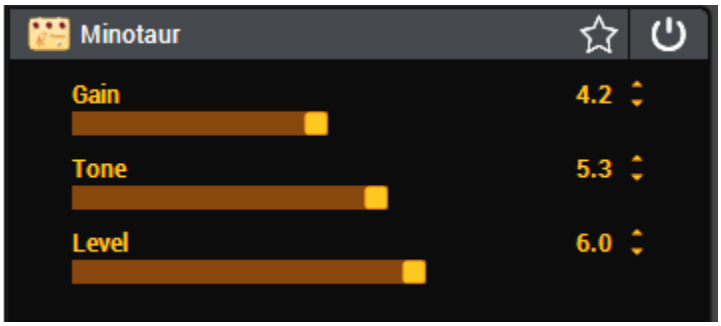
There aren't detailed descriptions of Distortion blocks with common parameters like Drive, Tone, Bass, Treble, Level, and the like, because most players already know how those parameters affect the sound. So, this section prioritizes distortions that have unusual parameters or functionality. Parameters like "Grunt" or "Pull Diode" *do* require explanations.

However, before getting into the complex distortions, let's analyze the 3-parameter distortion blocks. All of them have tone controls that modify the distortion character. Analyzing the tone control curves may seem irrelevant, because your ears make the important decisions. But understanding the response curves may help you narrow down your choices for which distortions have a tonal quality that would work well with specific presets. After that, choosing the right distortion involved auditioning the different clipping characteristics—softer, harder, buzzier, smoother, etc. I hope you find looking over these curves as useful as I did when trying to figure out which distortion to use out of the many choices.

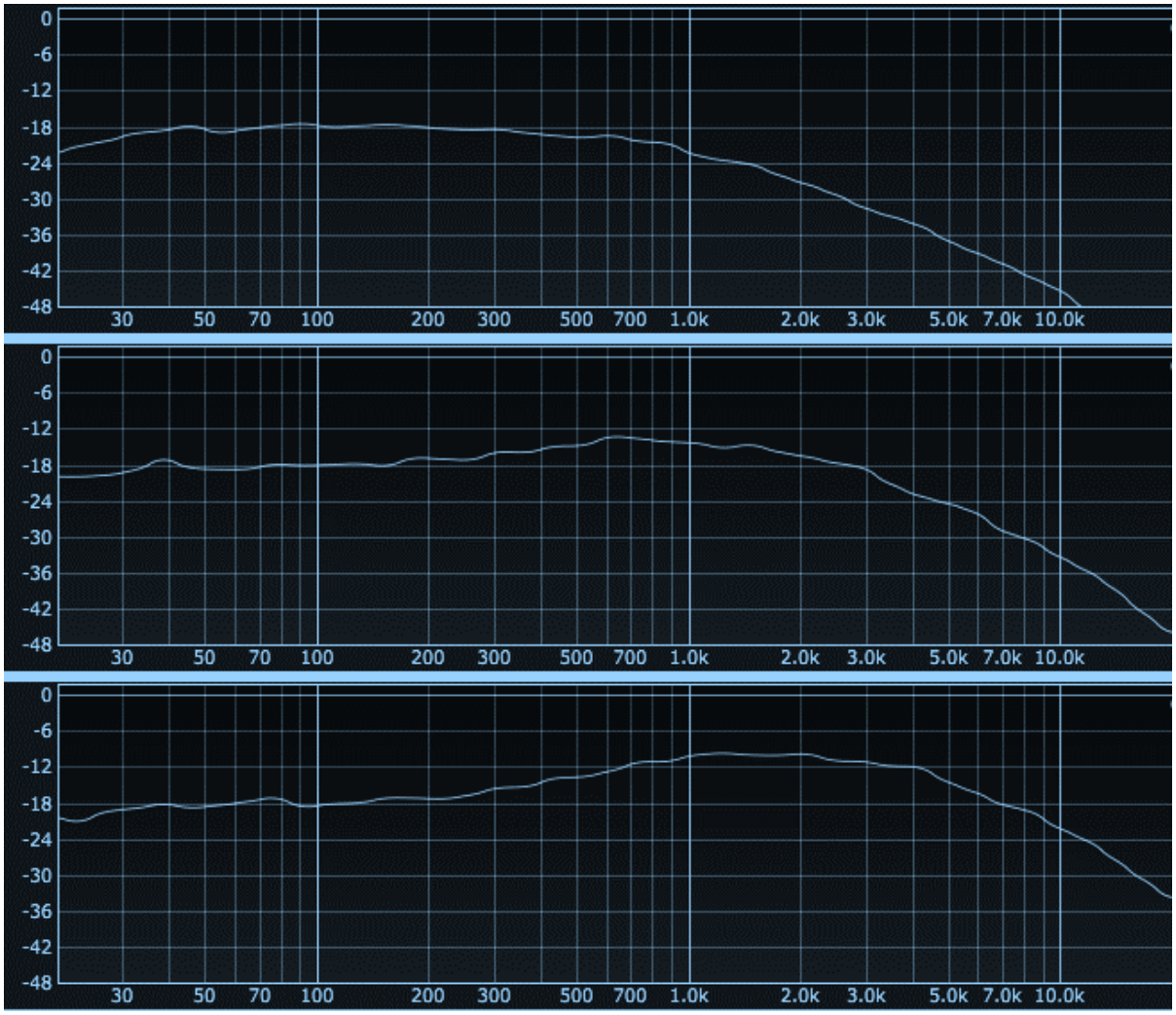
Distortion Blocks with 3 Parameters

In the following tone control frequency response graphics, the top graphic is the 0.0 tone setting, the middle graphic is the default tone setting (often, but not always, 5.0), and the lower graphic is the 10.0 setting.

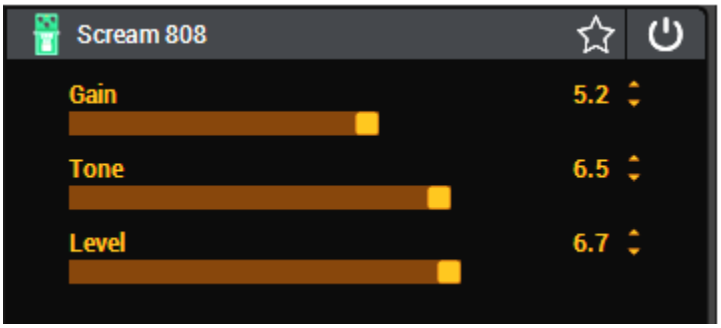
Minotaur



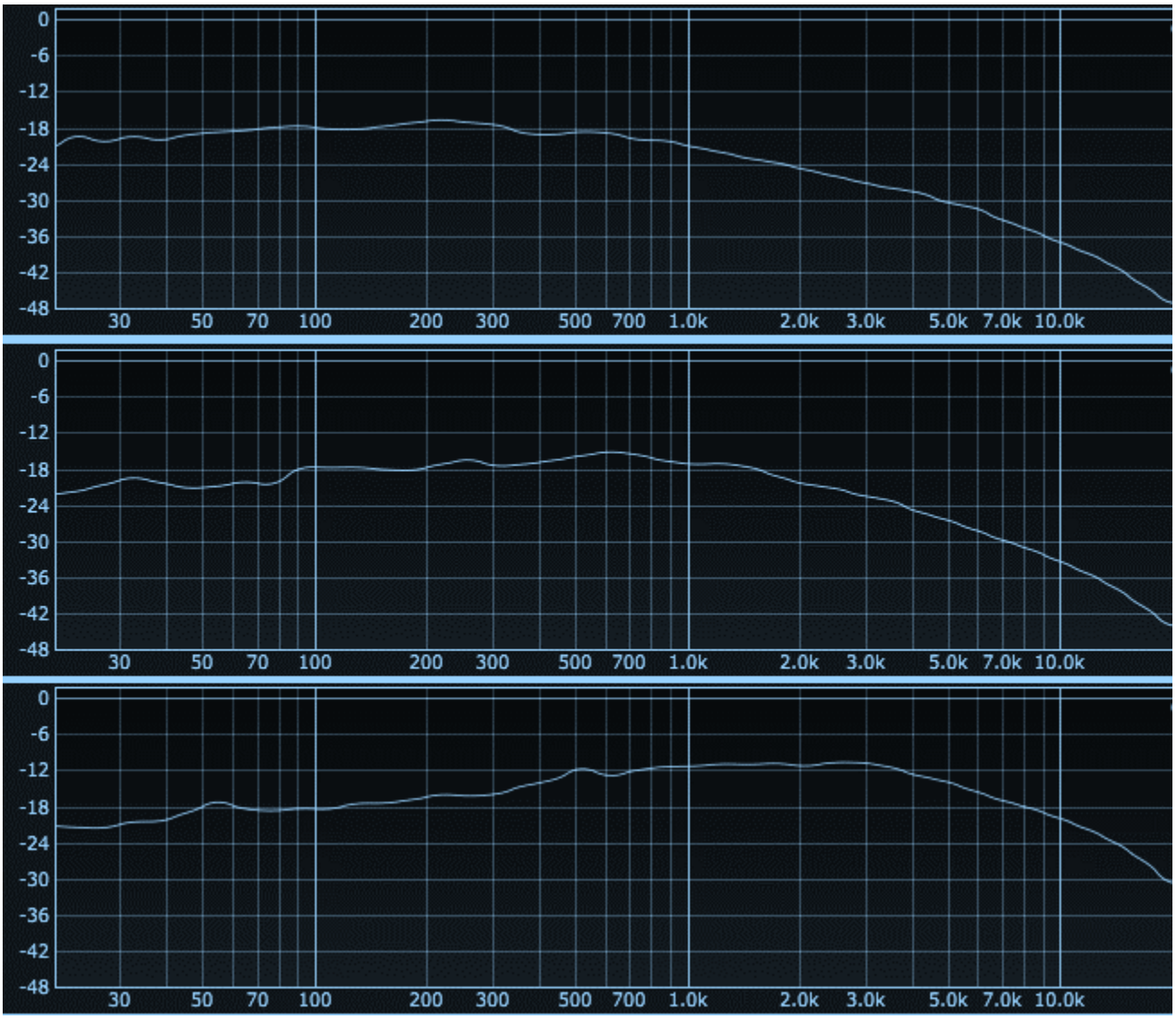
Many consider this soft-clipping fuzz as sounding “transparent” because the EQ doesn’t have a huge effect on the timbre. 0.0 rolls off around 500 Hz. The default is fairly flat until it rolls off starting around 1 kHz. 10.0 boosts the midrange. This doesn’t slam you with distortion and is good for rhythm guitar.



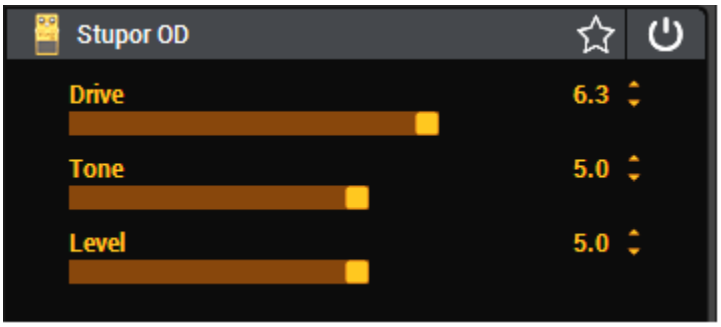
Scream 808



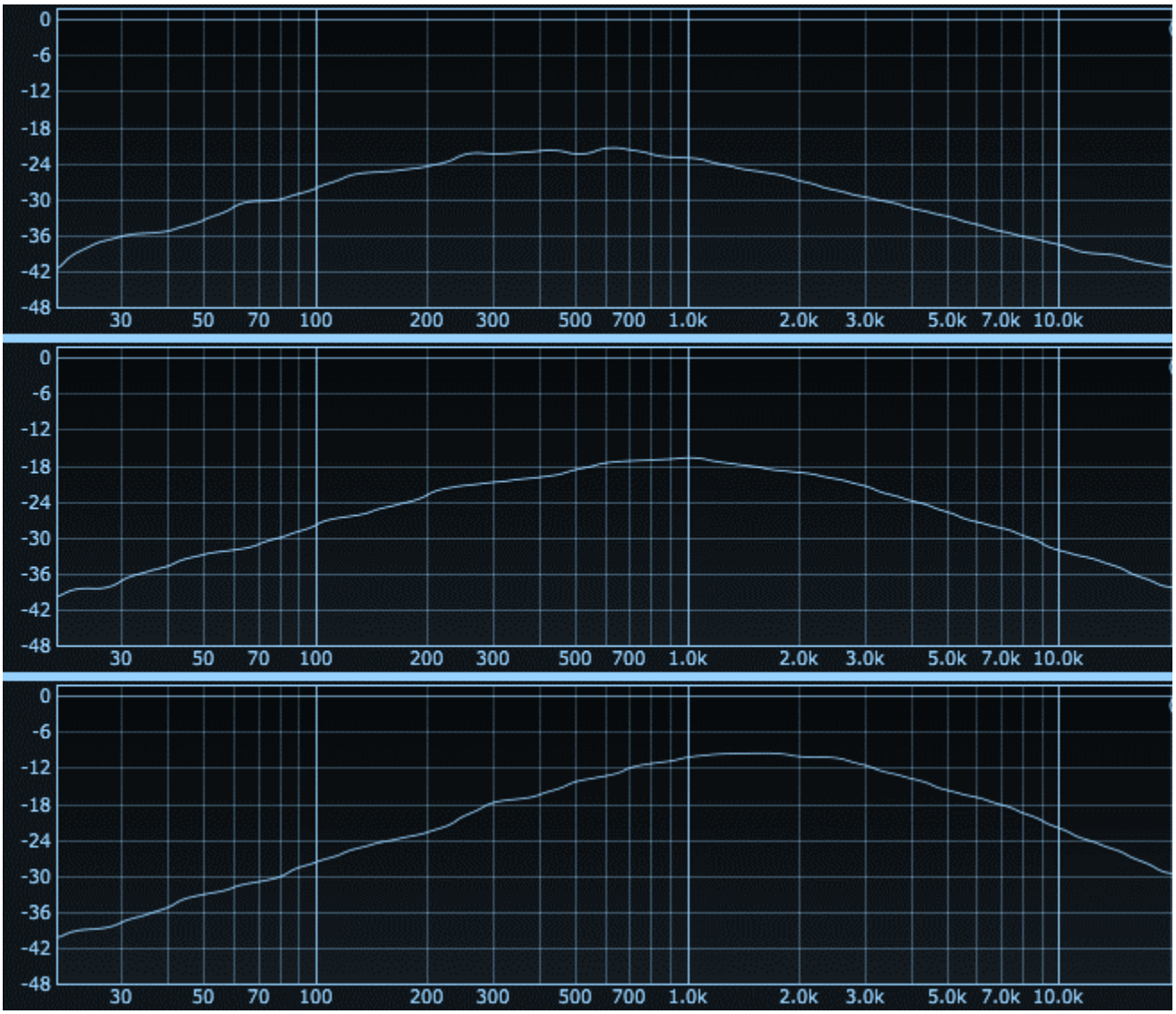
This shows why people often compare the Klon and Tube Screamer. They both have conservative tone shaping, and are good choices for rhythm overdrive. 0.0 rolls off highs around 500 Hz. Moving to the default rolls off the bass more and extends the midrange. 10.0 gives the Screamer's distinctive midrange peak and rolls off bass around 1 kHz. This setting gives the Scream 808 good definition, no mud, and no excessive highs.



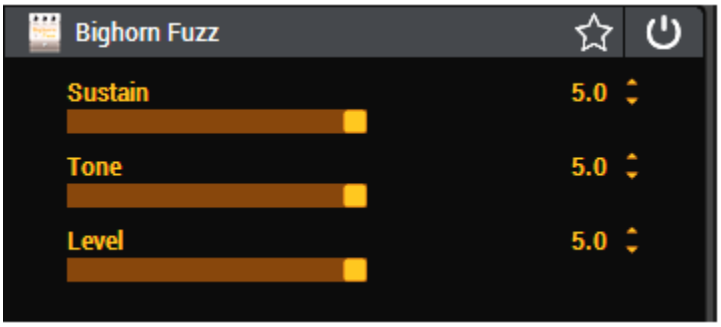
Stupor OD



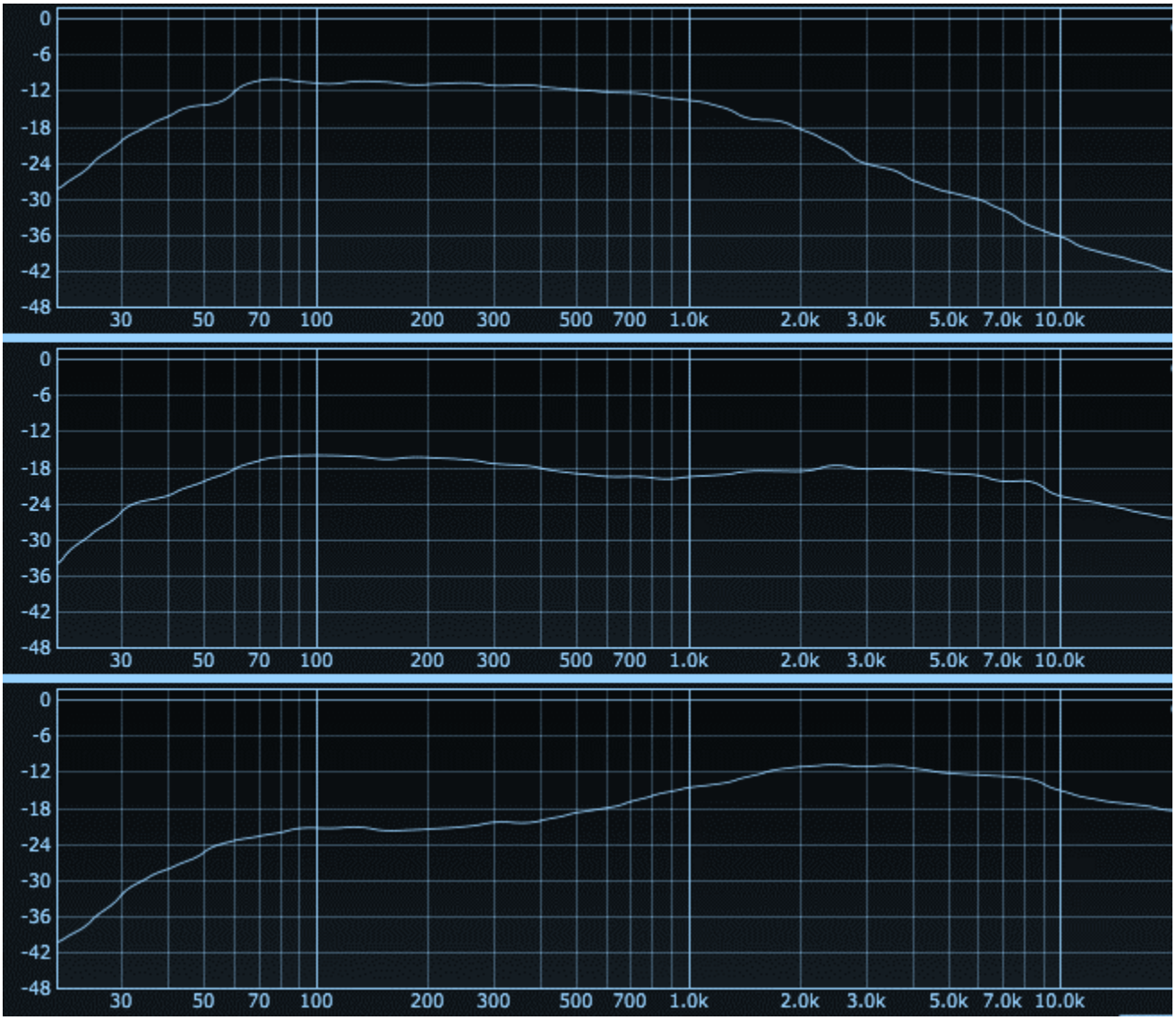
This overdrive takes the Tube Screamer's "boost the treble, drop the lows" to a greater degree. 0.0 has a broad, moderate boost in the lower mids that falls off at the high and low frequencies. The default moves this broad peak up to 1 kHz and adds some boost. 10.0 moves the broad peak higher, up to the 1 - 2 kHz range, and delivers the most amount of boost. The result is a sweet high end and reduced low-frequency "meat."



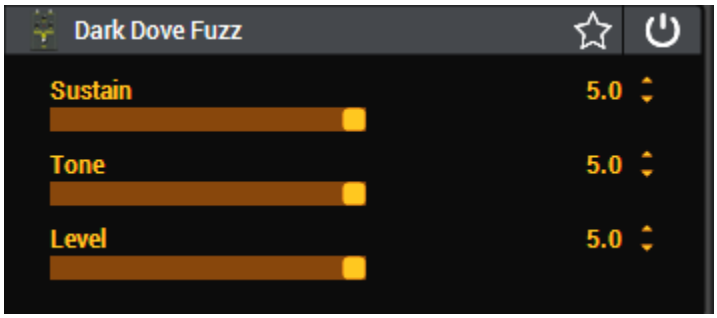
Bighorn Fuzz



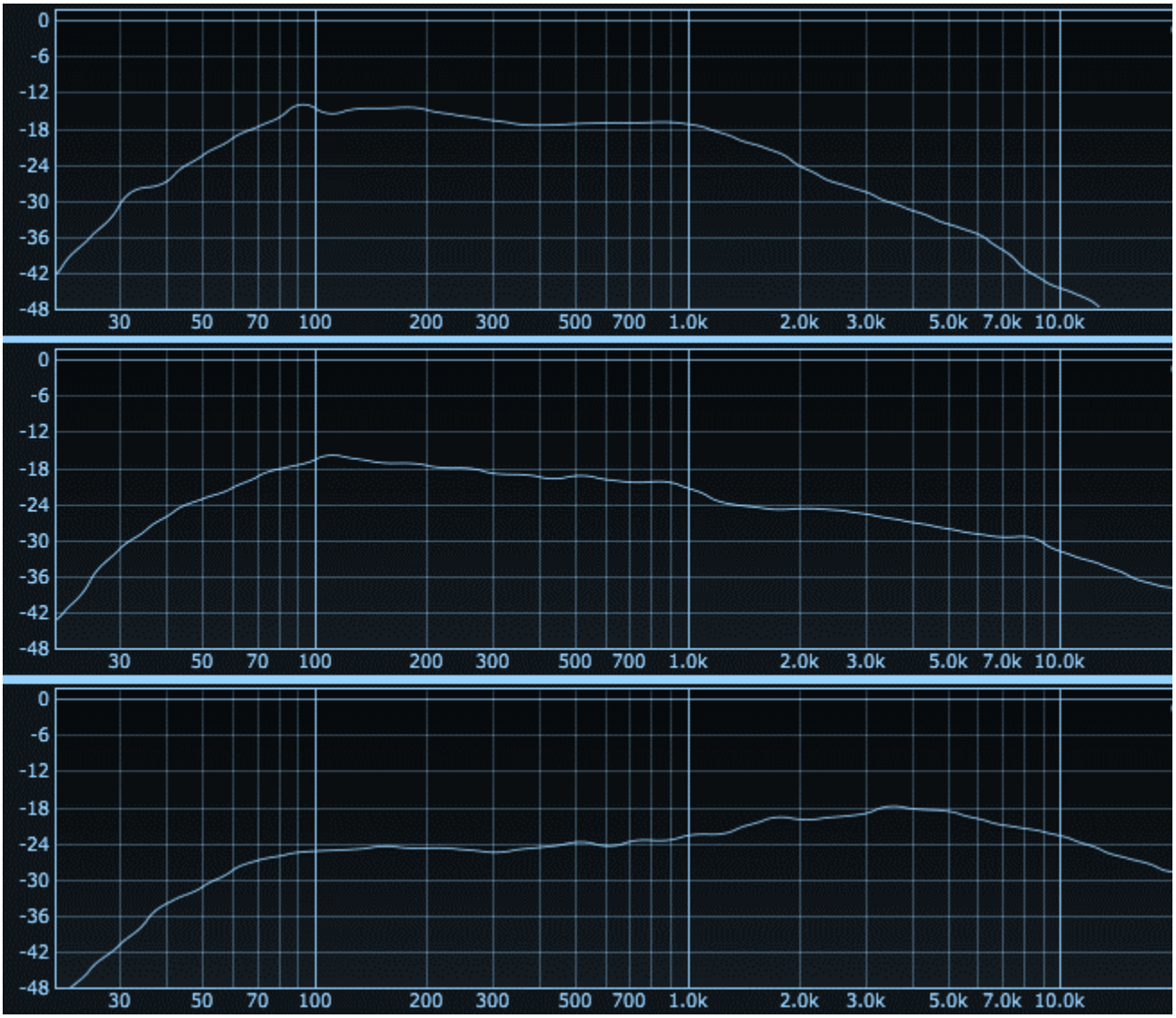
This is the first in a family of fuzzes: Bighorn, Triangle Fuzz, and Dark Dove are all related to the EH Big Muff. The star here is fatness with lower tone settings. 0.0 gives a major bass boost until it starts to roll off at 500 Hz. The default setting has small bumps at 150 Hz and 3 kHz. 10.0 boosts the mids around 3 kHz. At least to my ears, the highs are a bit harsh, so I tend to pull back the Tone a bit.



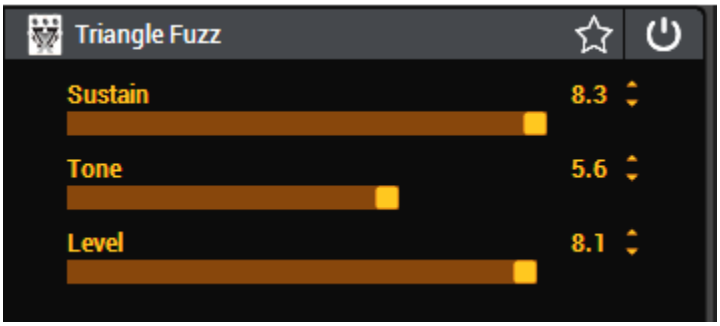
Dark Dove



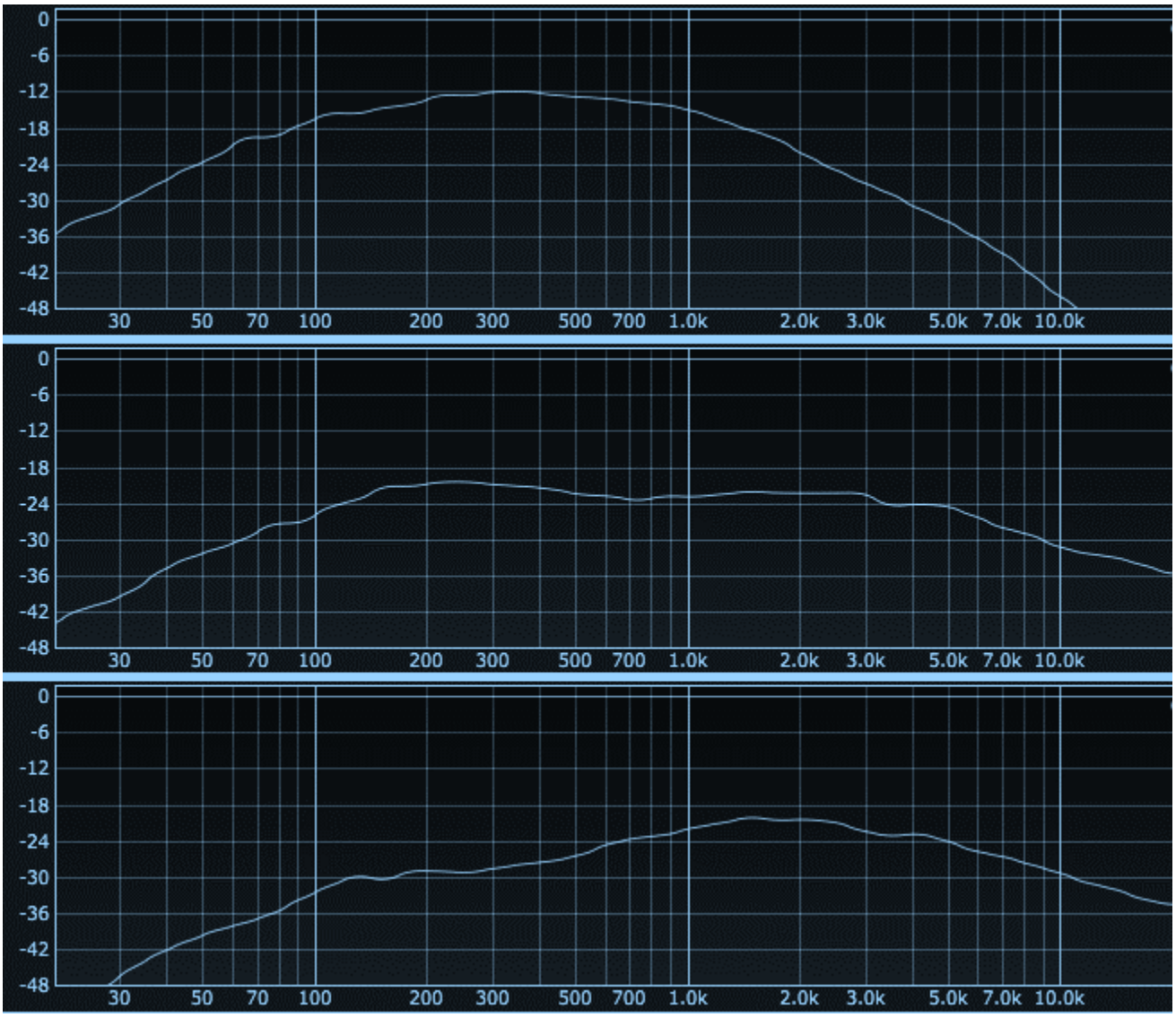
This has a similar personality to the Bighorn but is more about a big low end. It doesn't boost the highs as much at 10.0, and maintains a lot of low frequencies in the default setting.



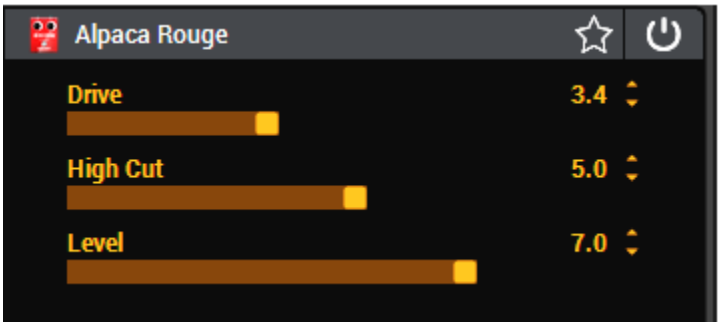
Triangle Fuzz



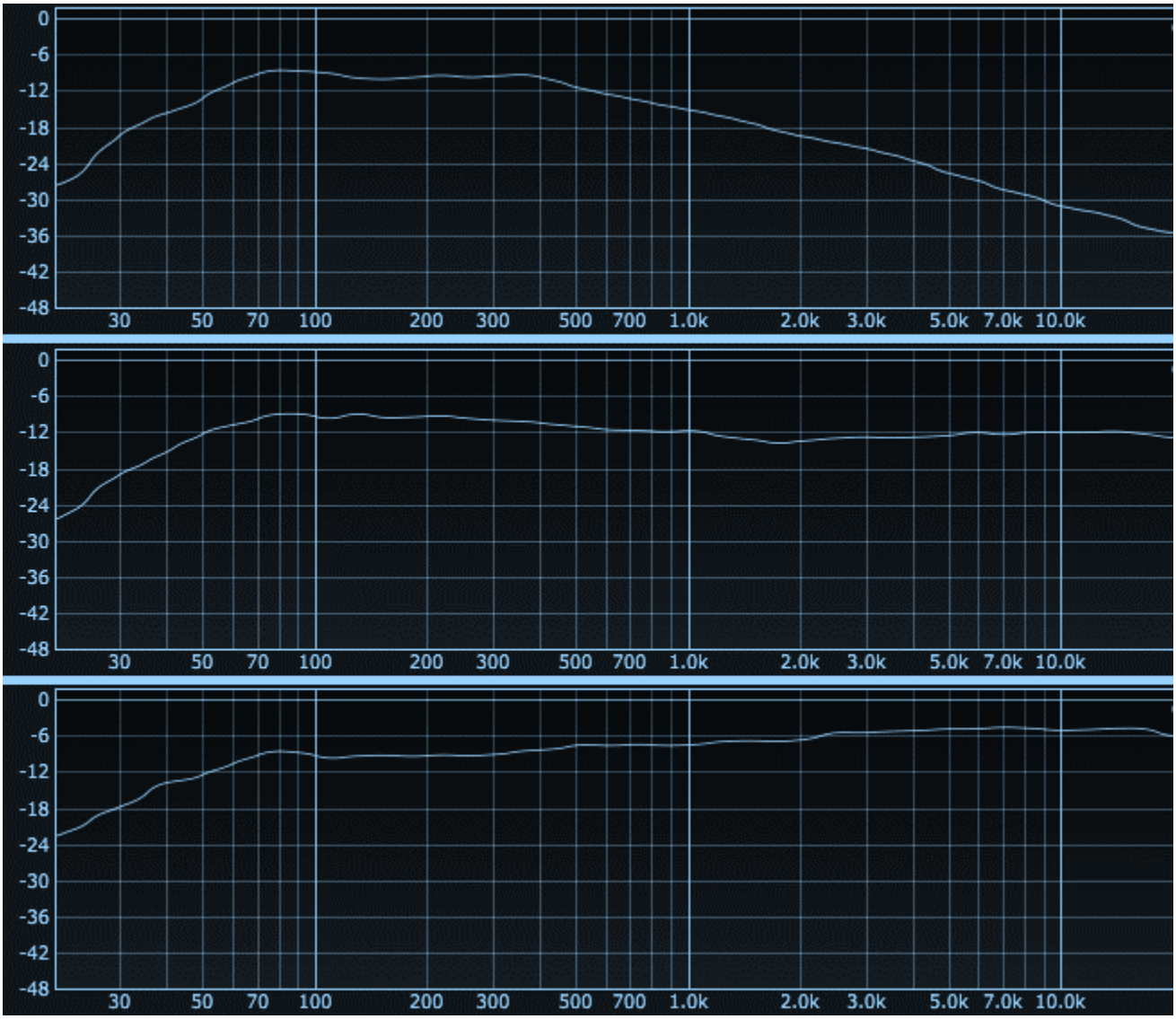
This is closer to the Bighorn than the Dark Dove. The main difference is that the Triangle Fuzz is more bass-shy overall, has a broad, mild boost around 300 Hz, and doesn't have as much treble or level at 10.00.



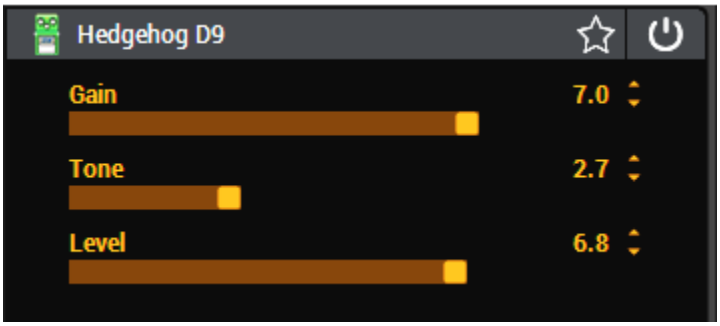
Alpaca Rouge



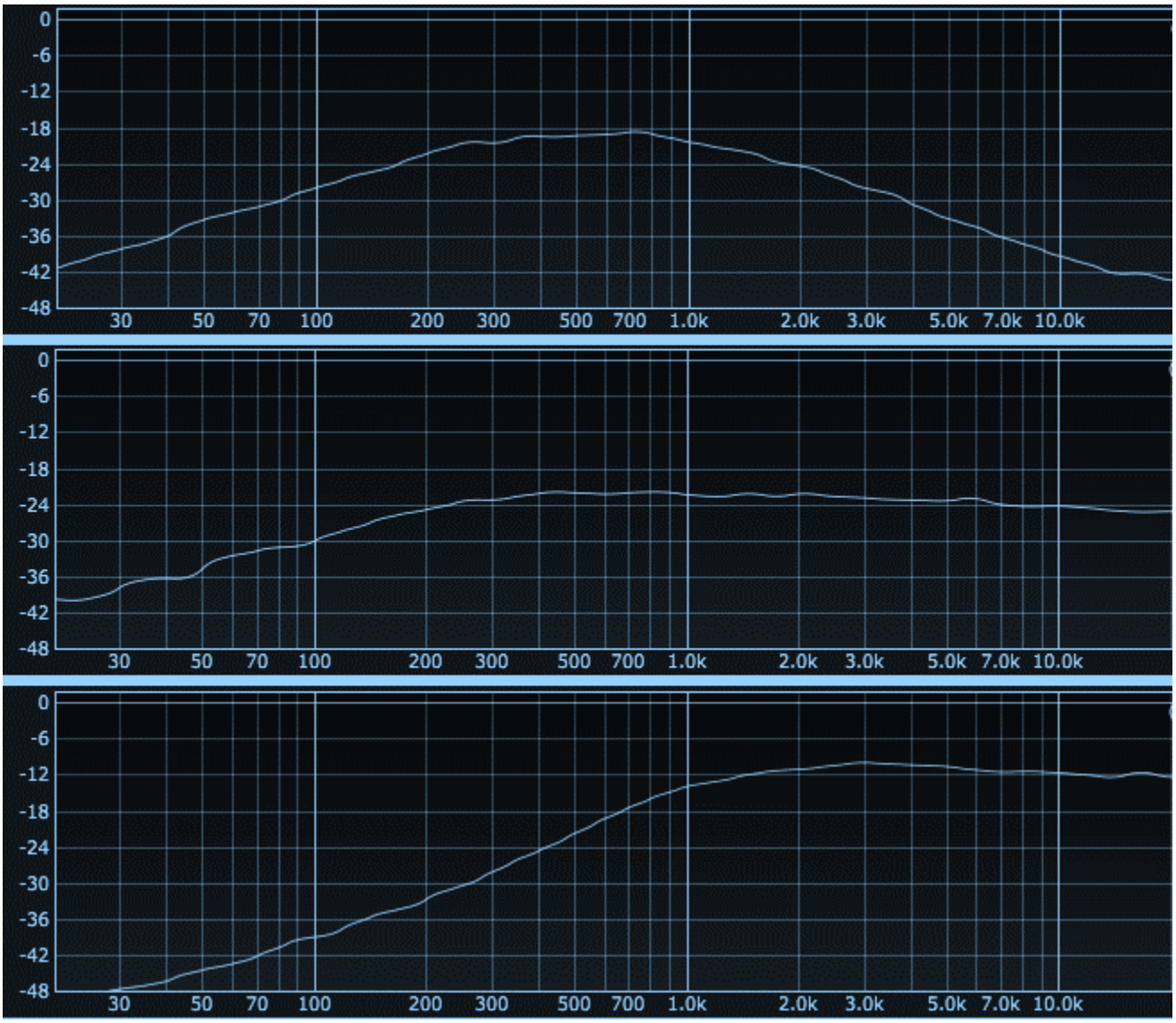
This is based on my Tube Sound Fuzz design from 1975, which has inspired many clones and mods over the years. It wasn't really a fuzz or overdrive in the traditional sense, but more like a tube emulator which became more guitar-friendly when overdriven. The main difference here is the addition of the High Cut control. At 0.0, the response rolls off the highs starting around 400 Hz. The response is almost flat at the default. At 10.0, there's a high-end boost and when run full-out, the Alpaca Rouge produces lots of output—it's very happy when overloading a tube preamp.



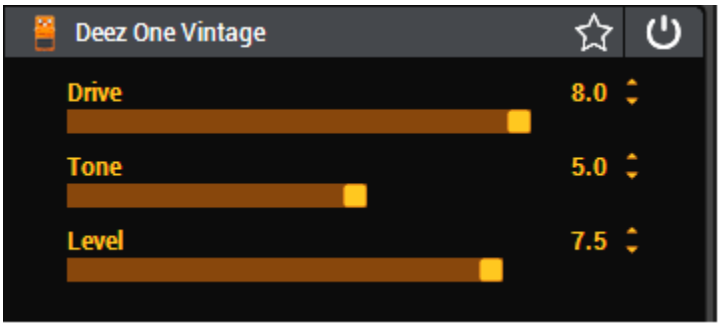
Hedgehog D9



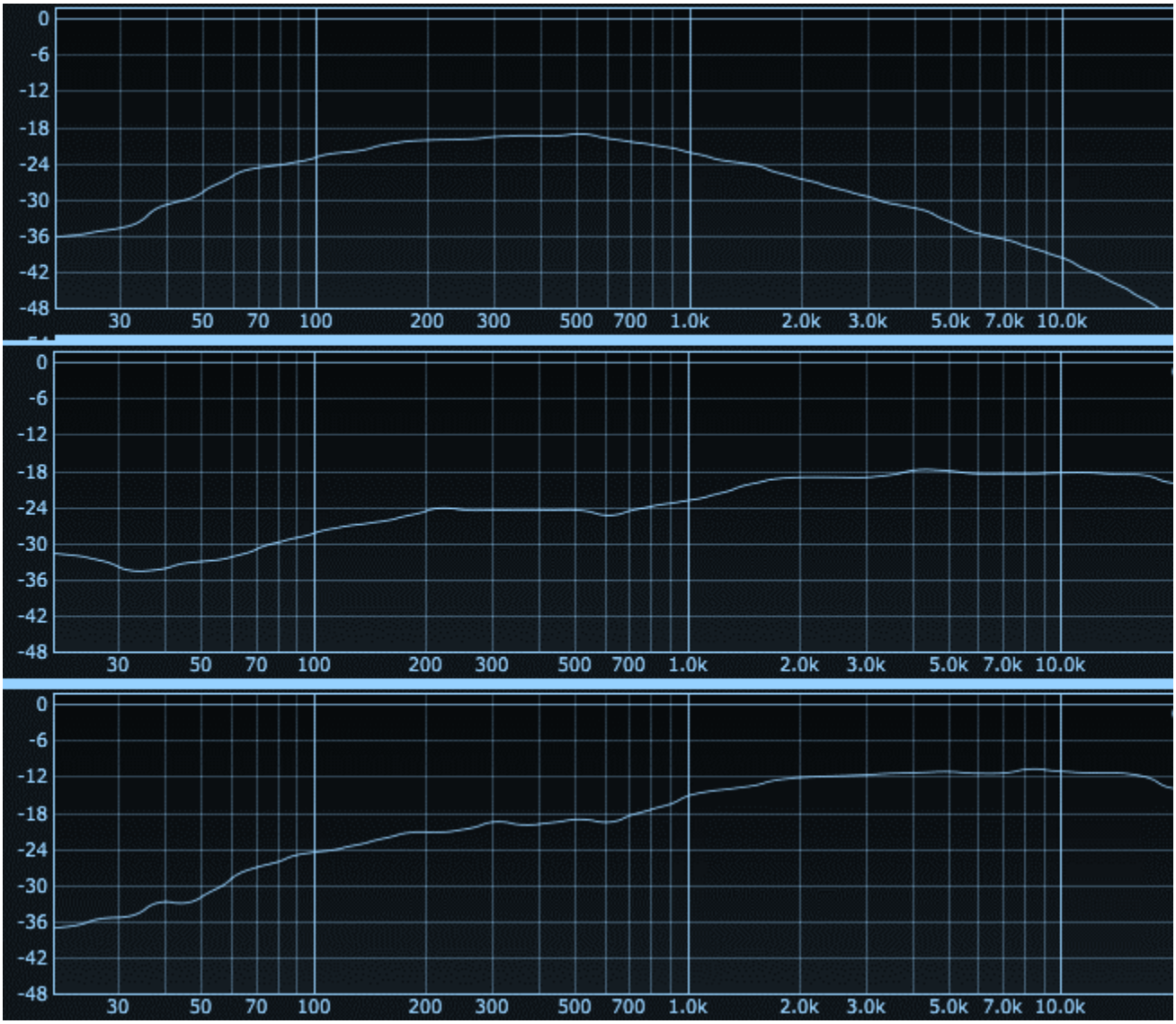
The main distortion characteristic is a smooth sound, from mild overdriven sounds to just short of metal. However, the Tone parameter is unusual, with what seems like a deliberate design. At 0.0, there's the broad midrange boost you see in many distortion devices. At the default, the response flattens out around 400 Hz, with a bass rolloff below that. At 10.0, the upper midrange emphasis makes for a defined, articulated sound with leads, while the steep bass rolloff makes room for other midrange instruments. Moving from 0.0 to 10.0 is like going from rhythm to lead.



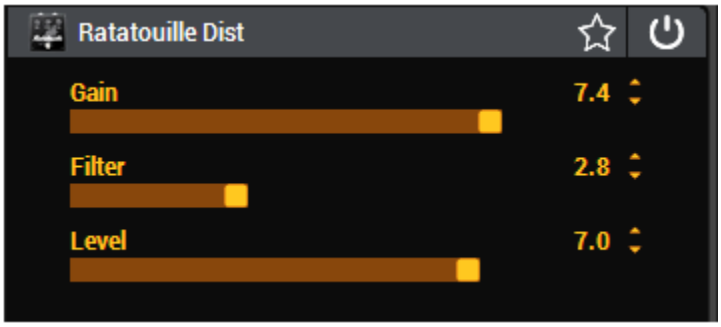
Deez One Vintage



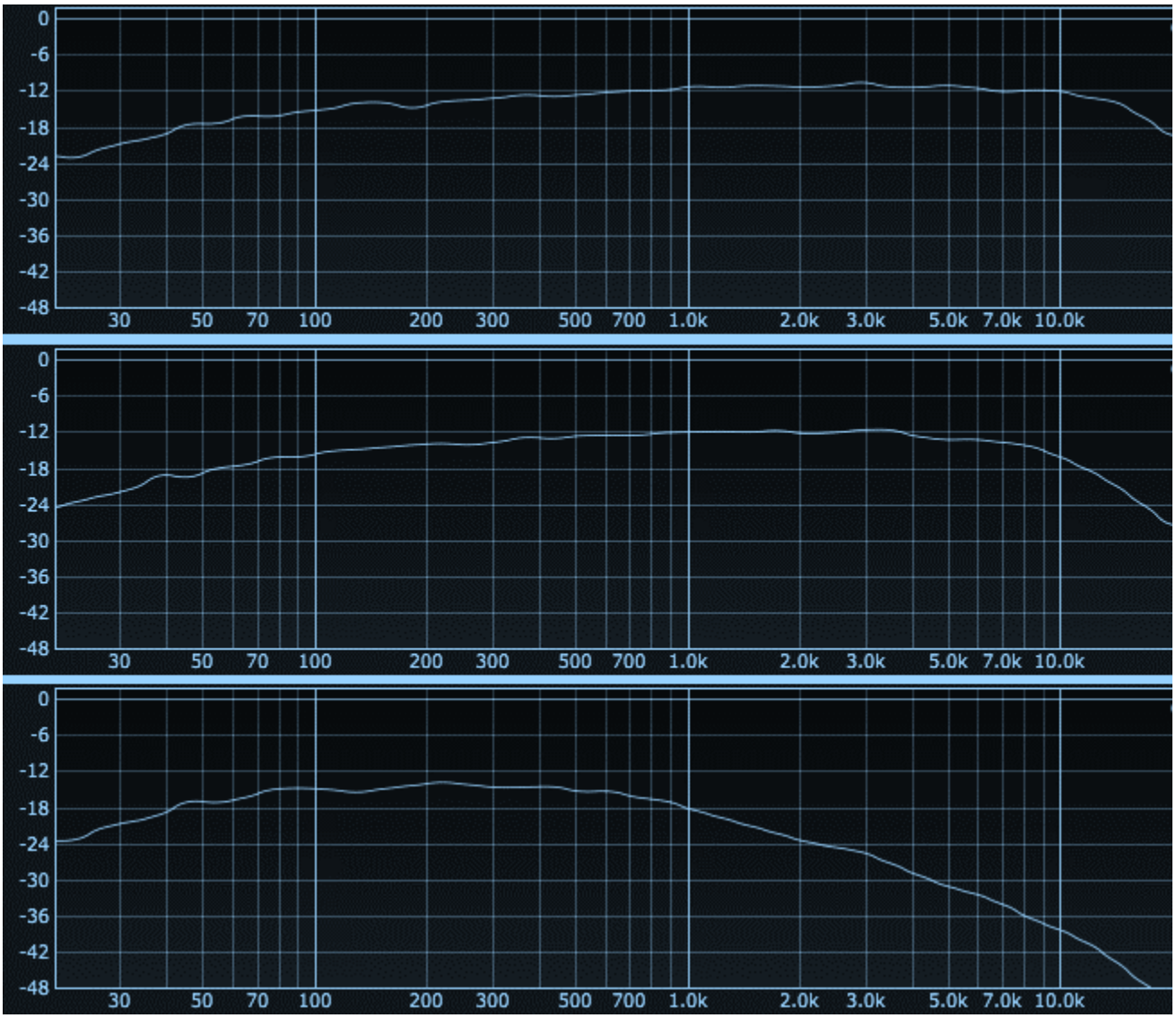
This is the poster child for a distortion stompbox tone control. There's a broad midrange boost at 0.0. Turning up to the default setting starts to raise the highs and lower the bass. Turning all the way up to 10.0 gives almost a tilt response from minimum bass to maximum highs. The overall effect is also rather gentle. To my ears, this helps compensate for some high-frequency brittleness.



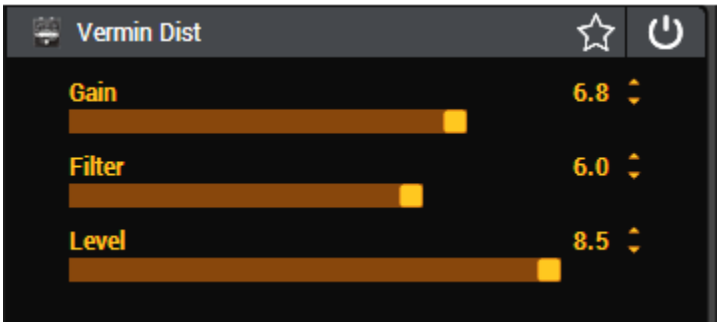
Ratatouille Distortion



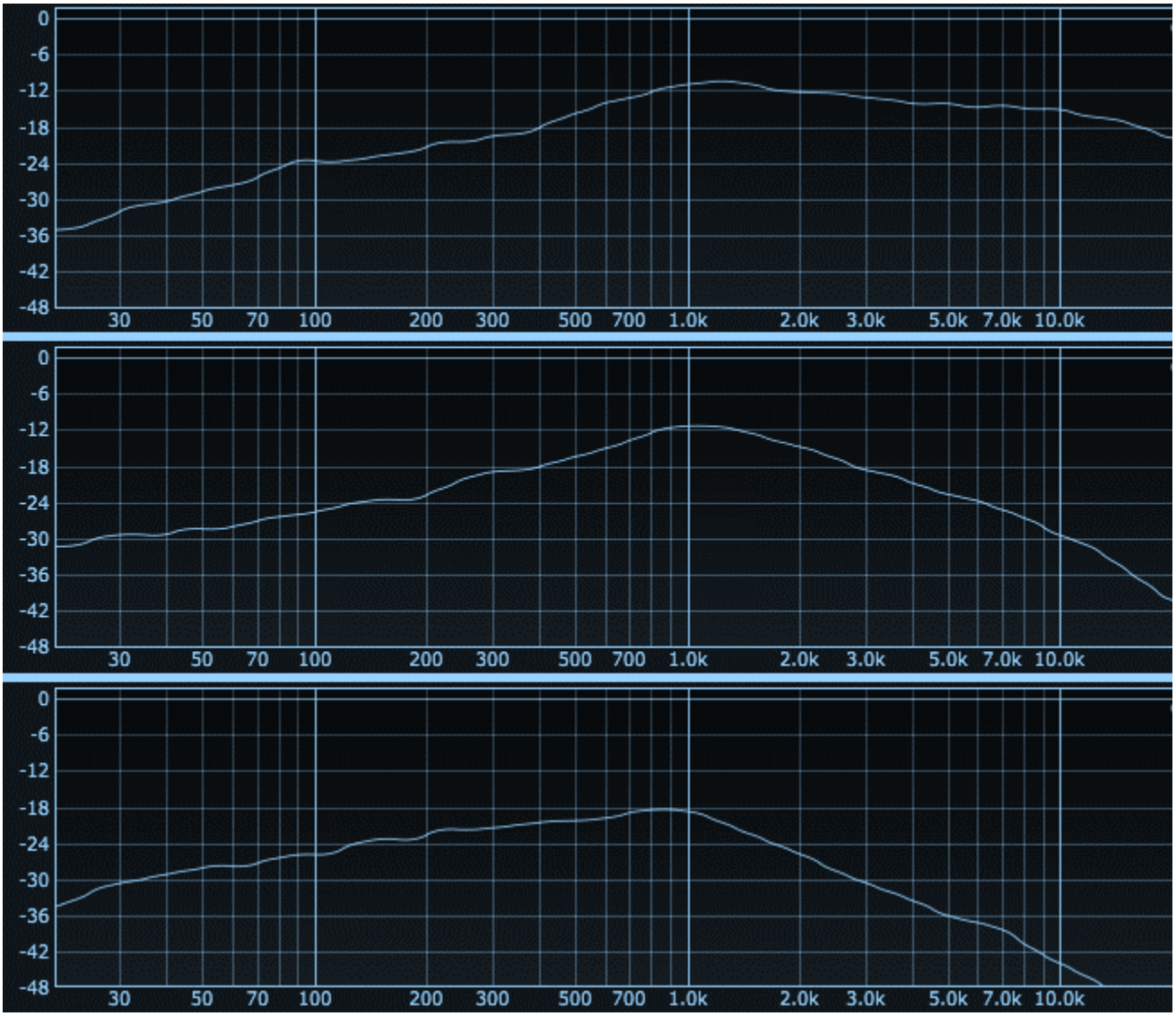
The Ratatouille's tone mostly involves controlling the high end, as the thick bass sound doesn't change much. 0.0 is close to flat, except for a slight tilt from bass to treble. The default setting doesn't make much difference, except the highs roll off faster. But at 10.0, the tone is all about pulling back hard on the high end, with a major rolloff coming into play around 400 Hz. It's somewhat unusual to see a distortion where turning up tone cuts the highs, but it makes sense for taming the otherwise potentially strident tone.



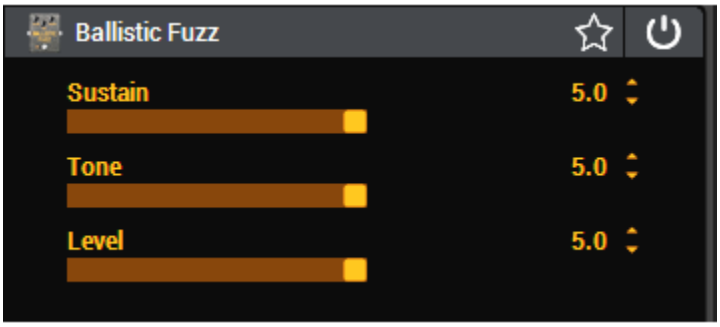
Vermin Distortion



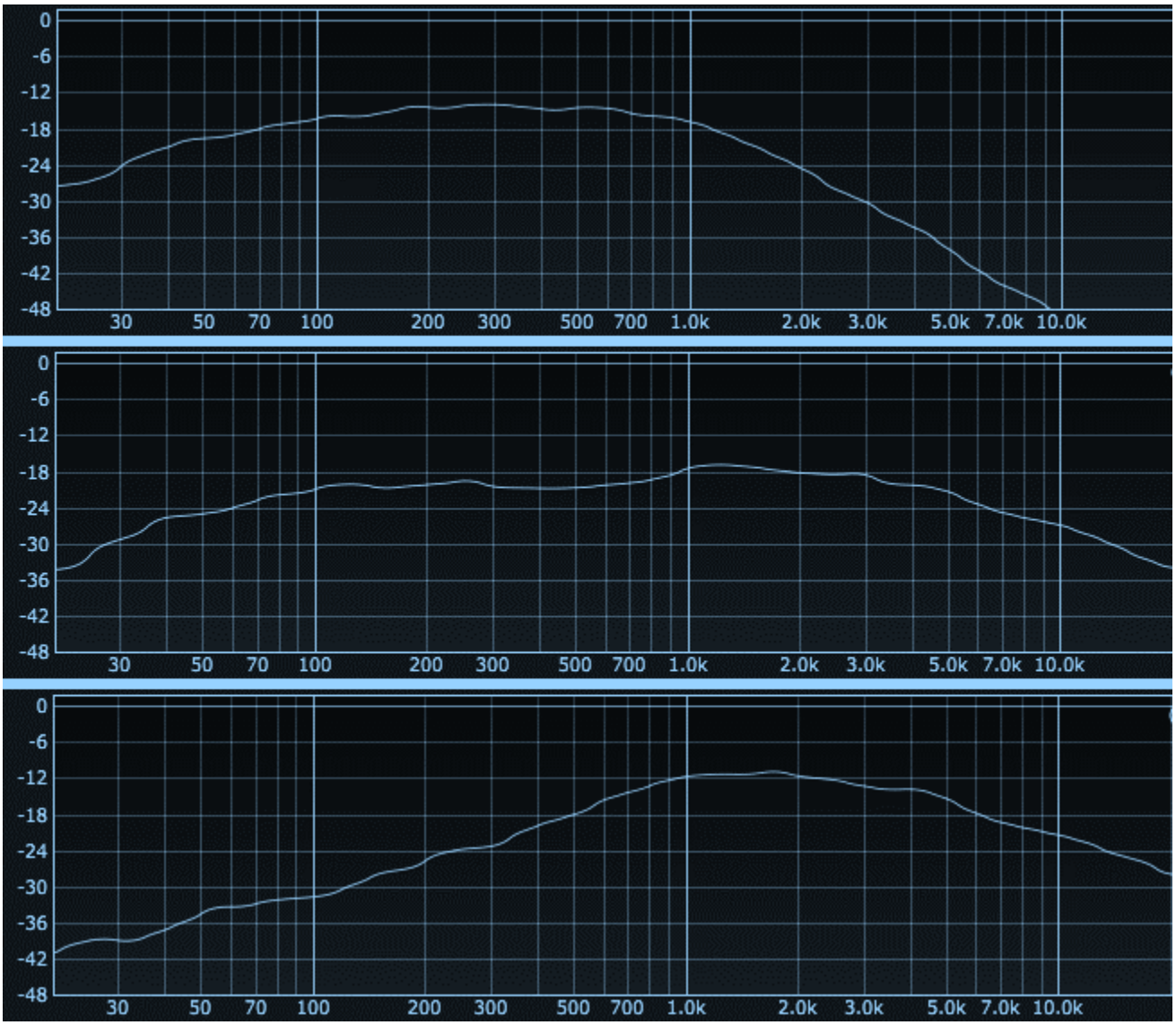
This has the most unusual filter effect of any of the distortions. The Vermin was supposed to model the Rat, but Line 6 unknowingly modeled a broken unit. The Ratatouille is a more accurate emulation, but the Vermin is a cool outlier with a truly unusual tone control. At 0.0, there's not a lot of bass, and mild high-frequency rolloff starting around 1 kHz. At the default, the significant peak at 1 kHz is almost like a mild wah, with rolloffs on either side. 10.0 brings back some of the bass and rolls off the highs more steeply.



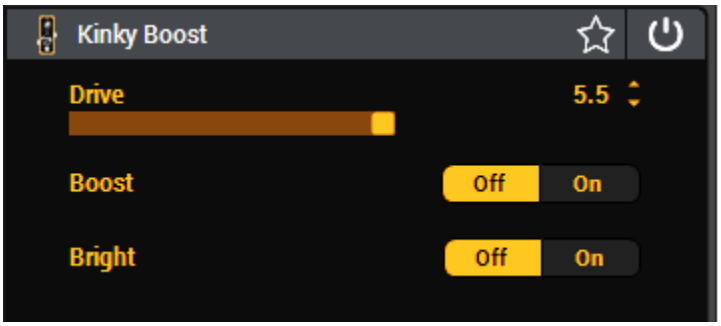
Ballistic Fuzz



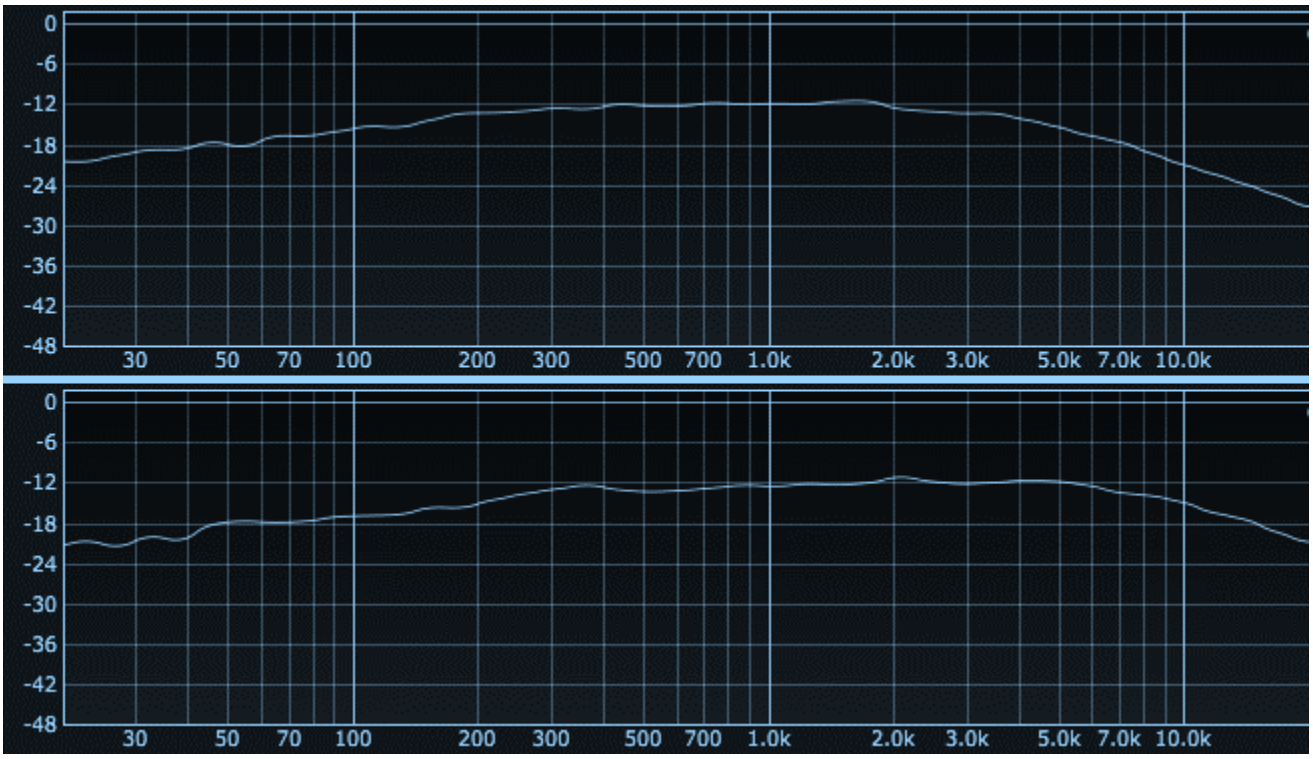
This distortion has a sharp, spiky character, so it's not surprising that the tone is about taming the high end. At 0.0, instead of the usual midrange boost, the high-frequency rolloff is much steeper than the low-frequency one. Tuning up to the default starts bringing the highs back in and drops the lower midrange a bit. 10.0 is almost like a mirror image of the 0.0 setting—a sharp rolloff below 1 kHz, a gentler high-frequency rolloff, and a broad peak at 1 - 2 kHz. Interestingly, at no point is there a high-frequency boost.



Kinky Boost



This really is just about a boost. There's not much going on, except the Bright switch makes it…brighter. When off, the highs roll off around 2 kHz. Enabling Bright moves the rolloff a little under 2 octaves higher. This model is simple enough that you can think of it as a palate cleanser before the entrée of complex models arrives.



Distortion Blocks with 2 or 4 Parameters

- Tycoctavia generates an octave-higher effect. For more information on how to get the most out of it, please see Taming the Tycoctavia. For another way to obtain octave-higher distortion, please see information about the preset 3B.+OctaveUp.
- The Swedish Chainsaw distortion's design focuses on heavy metal tones.
- The Ampeg Scrambler is optimized to be a bass overdrive.

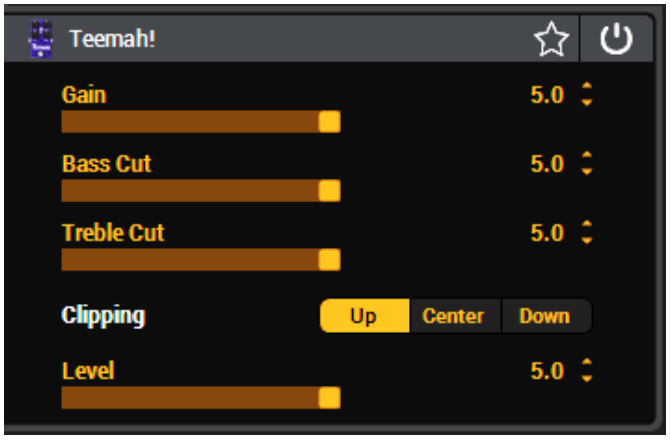
Top Secret OD
Gain 6.5
Level 8.3
Arbitrator Fuzz
Fuzz 9.0
Level 5.5
Pocket Fuzz
Drive 4.9
Level 4.5
Tycoctavia Fuzz
Fuzz 7.5
Level 6.7
Deranged Master
Drive 7.7
Bass 5.2
Treble 6.5
Level 0.0 dB
Dhyana Drive
Gain 4.4
Voice 5.6
Tone 4.4
Level 4.0
Valve Driver
Gain 6.5
Bass 5.5
Treble 4.5
Level 6.6
Deez One Mod
Drive 6.3
Tone 6.2
Level 2.5
Clipping
Symmetric
Asymmetric
Swedish Chainsaw
Drive 6.2
Bass 5.0
Treble 7.0
Level 6.0
Ampeg Scrambler
Drive 7.6
Blend 6.3
Treble 4.2
Level 6.6

Distortion Blocks with More than 6 Parameters

- The Horizon Drive's Gate section, which is more like a noise filter than a traditional gate, is available as a separate block called Horizon Gate.
- Legendary Drive is more accurately a preamp than a type of distortion. However, it's well-suited for running into a cleaner amp, to give serious overdrive.
- The Megaphone is…well, a megaphone. Use it with voice when you want to make an announcement, like "Your attention please—delayed flight number 4604 from Los Angeles is now arriving at gate B15." **Grit** adds distortion, **Focus** controls the aperture of the megaphone's horn, and **Space** adds an echo. For guitar, it's sort of like a resonator guitar, but less metallic, and resonates at a lower frequency. The Resonator in the Favorites folder gives a more convincing resonator guitar effect.
- The ZeroAmp Bass DI is a bass driver. Turning up the **Drive** control can add tasty saturation, as mixed in with the **Blend** parameter. The sound is cleaner than a pushed, overdriven amp.

Teemah!

Placed before an amp, Teemah! brings out different colors from your amp and guitar, rather than creates heavy distortion. The effect is more subtle compared to most distortions. For example, the Clipping parameter is more about tweaking the character rather than transforming the sound.



- **Bass Cut** is minimal. It's noticeable mostly at higher gain levels.
- **Treble Cut** makes a more dramatic difference, at any gain level.
- **Clipping Up** has the densest sound, due to lots of harmonics (fig. 2.1).
- **Clipping Center** has a focused, spikier distortion sound, and less dense harmonic content.
- **Clipping Down** has the least amount of clipping and harmonic content.

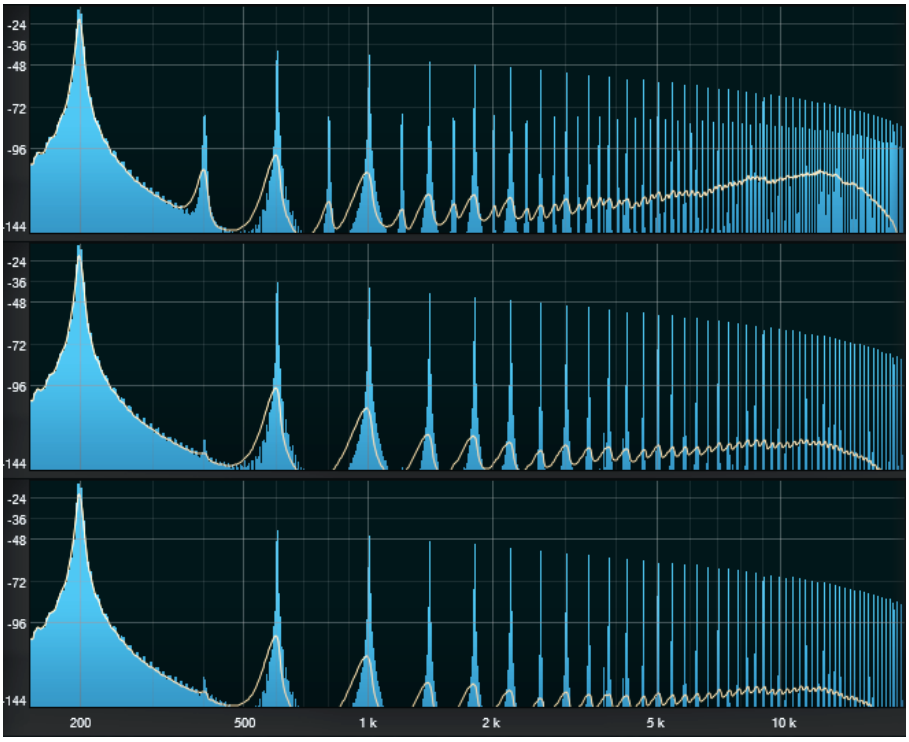
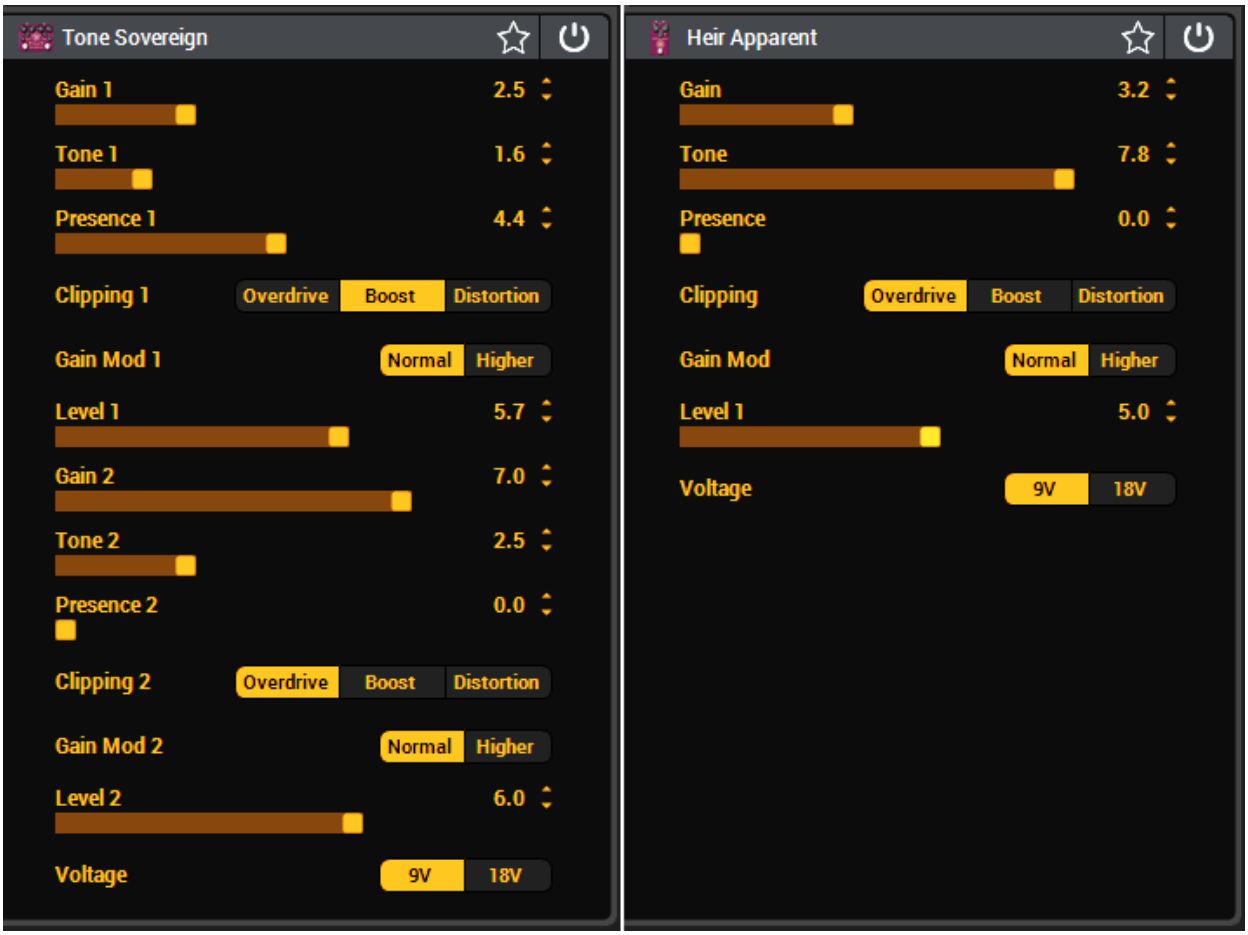


Figure 2.1 Top to bottom: Harmonics for Up, Center, and Down positions, with no Bass or Treble Cut, and medium gain.

Tone Sovereign and Heir Apparent

Tone Sovereign is two Heir Apparent effects in series, with a common Voltage parameter. Tone Sovereign cannot split into two independent channels. Using two Heir Apparent blocks instead of a single Tone Sovereign provides two main options:

- Route them in series to enable one, the other, or stack them in series.
- For a traditional two-channel setup, create a parallel path, and insert an Heir Apparent in each path.



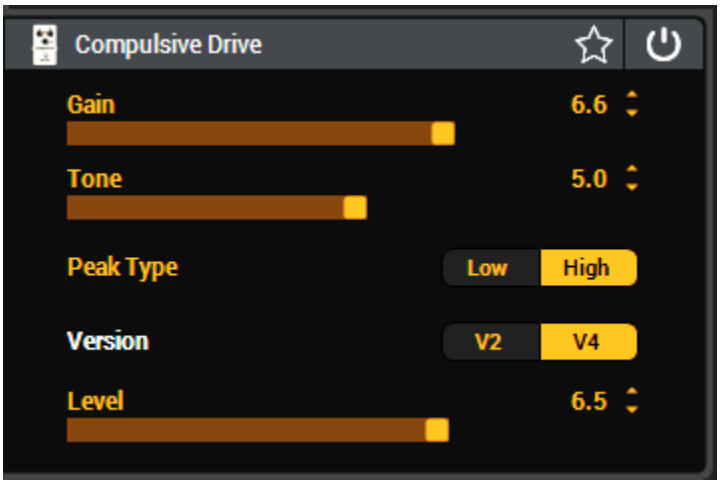
- **Overdrive**, **Boost**, and **Distortion** give moderate distortion, clean boost, and hard distortion.
- **Normal/Higher Gain Mod**. Higher Gain Mod is optimized for distortion.
- Low **Gain** settings clean up the sound well.
- The **Voltage** switch gives more headroom. This is useful with the Boost setting if you crank up the Gain.

Tip: For an ersatz Tone Sovereign two-channel workaround, set Gain 2 to 0.0, and adjust the parameters for Section 1 as desired. Then, set Gain 1 to 0.0, and adjust the parameters for Section 2 as desired.

Program a footswitch so that one position sets Gain 1 = 0.0 and Gain 2 for the setting you chose when Gain 1.0 was 0.0. Program the other footswitch position so Gain 2 = 0.0 and Gain 1 is the setting you chose when Gain 2 was 0.0. Note that because the settings between the two sections interact, this is a less flexible approach than using two Heir Apparents.

Compulsive Drive

The Compulsive Drive emulates two different versions of the OCD effect.



The **Gain**, **Tone**, and **Level** controls work as expected, but there are a couple twists:

- **Peak Type** controls a filter. Low is the least colored setting. High applies a very gentle midrange frequency boost, centered around 1-2 kHz.
- The **Version** chooses different distortion characteristics (fig. 2.2). V2 squares off the signal more than V4. V4's sound is rounder and warmer, while V2 is "spikier" and brighter.

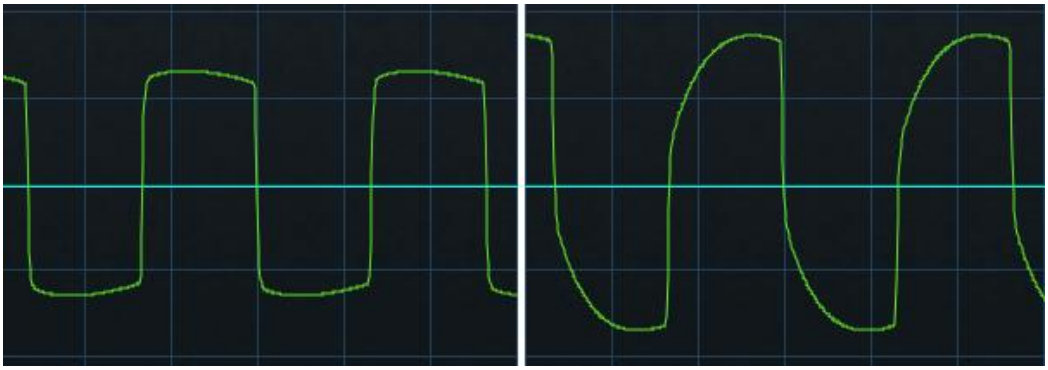


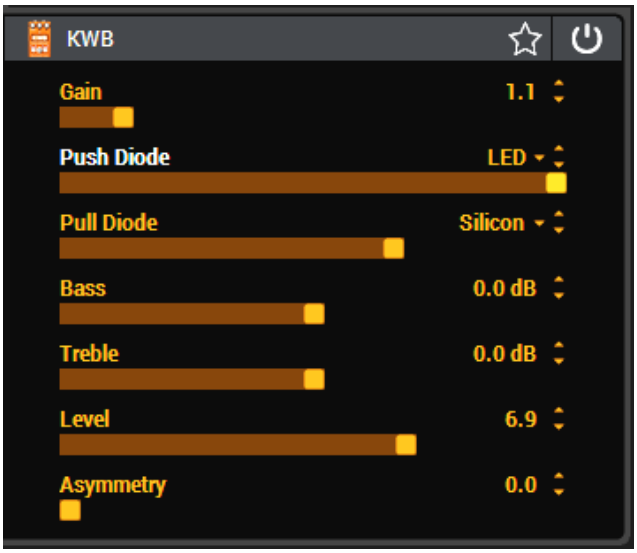
Figure 2.2 How V2 and V4 affect a sine wave.

Tip: The warmest sound uses Low Peak Type and V4. The most distorted sound uses High Peak Type and V2.

KWB Distortion

Most distortions are straightforward—then there's the KWB Distortion. This could just as easily have been named the HYVSI ("Here's Your Virtual Soldering Iron") Distortion.

Diodes are popular components for creating distortion, and different diode technologies have different distortion "characters." The KWB emulates the three different types of diodes that are used most often to create distortion.



Tech Talk: Diode Technologies

The KWB Distortion's "secret sauce" is the option to choose different diode technologies to clip the positive audio peaks (the Push diode) and negative peaks (the Pull diode) independently. Different diodes clip at different levels, and round off the edges differently (fig. 2.3) as they get hit with more level. To make the differences more apparent visually, the waveforms are shown with fairly low Gain. However, the diode characteristics do interact with the Gain control and Asymmetry parameter.

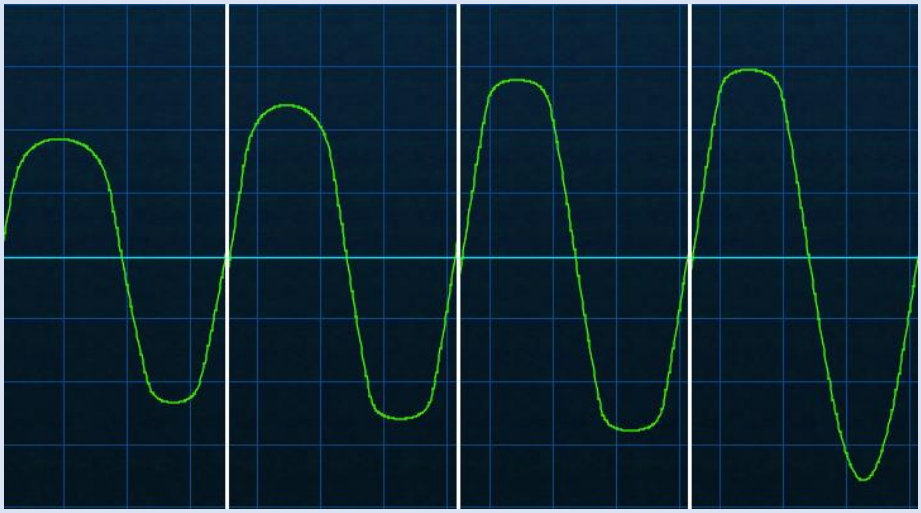


Figure 2.3 All waveforms shown use the LED Pull diode. Push diodes, left to right: Germanium, Silicon, LED, and LED with increased Asymmetry. Whether Asymmetry increases the positive or negative peak depends on the Gain, the selected Diodes, and the Asymmetry amount.

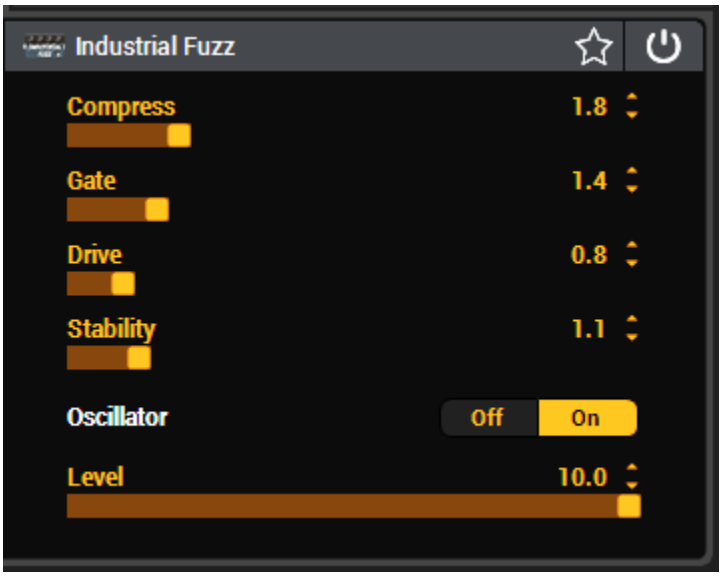
Sonically, here's how different diodes react:

- **Germanium** diodes clip at lower levels than other types of diodes. Even low Gain settings give fairly heavy distortion.
- **Silicon** diodes clip at higher levels. With moderate Gain settings, picking the strings harder gives heavier distortion.
- **LEDs** not only need higher levels to clip, but they also go into distortion more smoothly. This type of diode gives the most touch-sensitive distortion. Red LEDs are my favorite distortion element. (Boston's Tom Scholz is also a fan of red LEDs.) Other LED colors clip at a higher level and aren't commonly used.
- The **Asymmetry** control alters the distorted sound's "buzziness."
- Selecting **No** diode can give a crunchy, touch-sensitive overdrive.
- If you choose the *same* diode type (silicon, LED or germanium) for the Push and Pull diodes, with low Gain settings any sonic difference between the Push diode or Pull diode by itself is subtle. However, with higher Gain settings, the Pull diode has more level and slightly more low end than the Push diode.

Tip: Any differences matter most at low-to-moderate **Gain** settings. If you drive any diodes with high levels, they eventually become more like square waves, and won't sound as different.

Industrial Fuzz

If circuit-bending, distortion, and randomness had a kid, this would be it. Your guitar interacts with oscillations modeled within the effect. Some assume this effect is based on ring modulation, but it's not. The most interesting aspect is that Line 6 has been able to emulate analog unpredictability in a digital device.



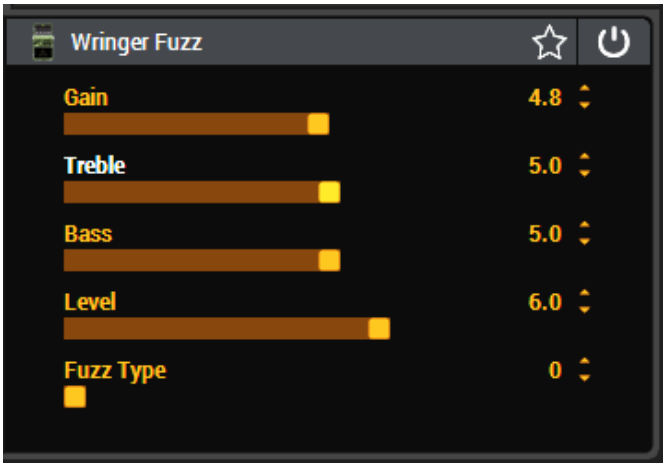
A device like this is difficult to explain, and the control names don't help—**Gate** is not a gate, and **Compress** isn't a compressor. Instead, the controls interact unpredictably, and interact equally unpredictably with guitar.

These tips will help you become familiar with this effect:

- Try following the Industrial Fuzz with a relatively low-distortion amp, like the US Double Nrm. This type of amp can help "tame" the effect, without detracting from the sonic mayhem. Adding more distortion after the Industrial Fuzz ends up sounding more like white noise, although of course, that might be what you want.
- The oscillation feature is far more dramatic with chords, because the interaction leads to lots of intermodulation distortion. To get a feel for the controls, set them as shown in the above screenshot, but don't play your guitar yet. Vary the controls, and you'll hear the oscillator change pitch.
- Start playing some power chords, and your guitar will interact with the oscillator. If it's not clear what the **Oscillator** parameter is adding, turn it off toward the end of a sustaining chord. This makes it easier to notice what's missing.
- Vary **Stability** and observe how it affects the sound's character.
- Higher parameter values can make the effect more predictable. Lower parameter values often introduce more chaos.

Wringer Fuzz

This emulates the modded version of a fuzz used by Garbage. The original device incorporated two fuzz sound variations in one pedal and a gain boost. This emulation omits the gain boost.



Fuzz Type switches between two post-fuzz filter networks. Both types add a smooth, octave-higher sound (I prefer it to the Tycoctavia). Regarding other characteristics:

- **Fuzz Type 0** is a vintage-sounding 60s effect, with a midrange emphasis (fig. 2.4). It has a more prominent octave-higher element.
- **Fuzz Type 1** scoops the mids at 1 kHz. The Treble parameter affects frequencies above 1 kHz. The Bass parameter affects frequencies below 1 kHz.

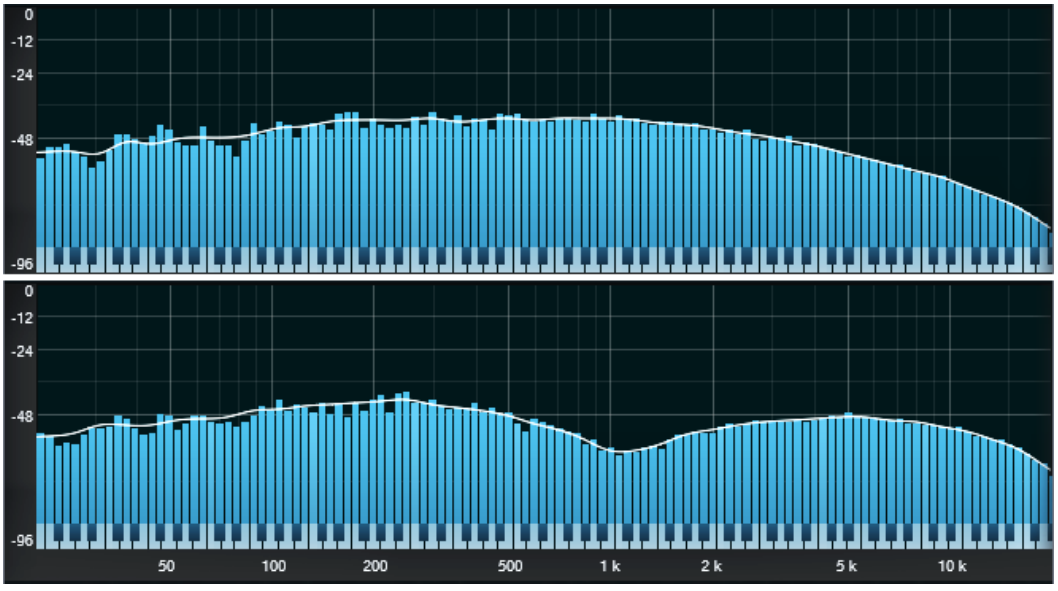
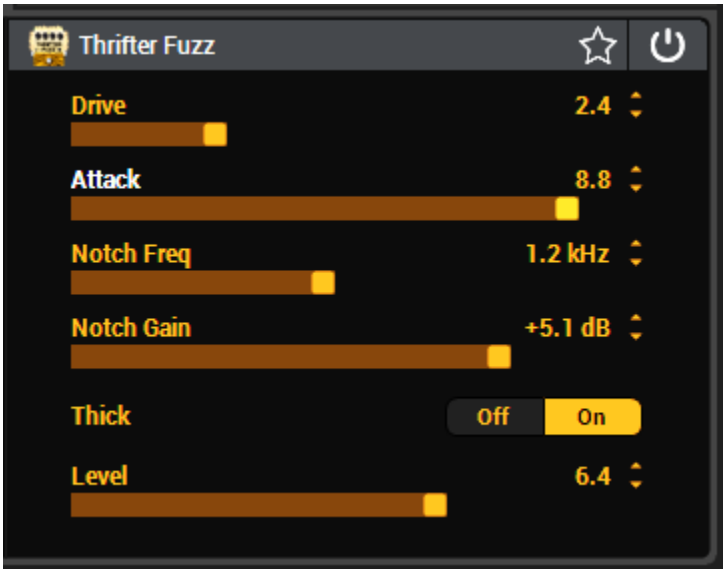


Figure 2.4 Frequency response for Type 0 (top) and Type 1 (bottom).

Tip: This fuzz works with chords, but single-note leads excel because of the octave-doubling effect. Try following the Wringer Fuzz with a cab, and use the Bass and Treble parameters to shape the tone. The 4x12 Cali V30 Cab works well with Fuzz Type 1 and a treble boost. The sound is different compared to using an Amp+Cab.

Thrifter Fuzz

Most Helix distortions emulate existing products. This Line 6 original design doesn't have a commercial precedent, and provides a different type of fuzz effect. Its main feature is including a filter to shape the tone. I prefer using this distortion with single-note leads.

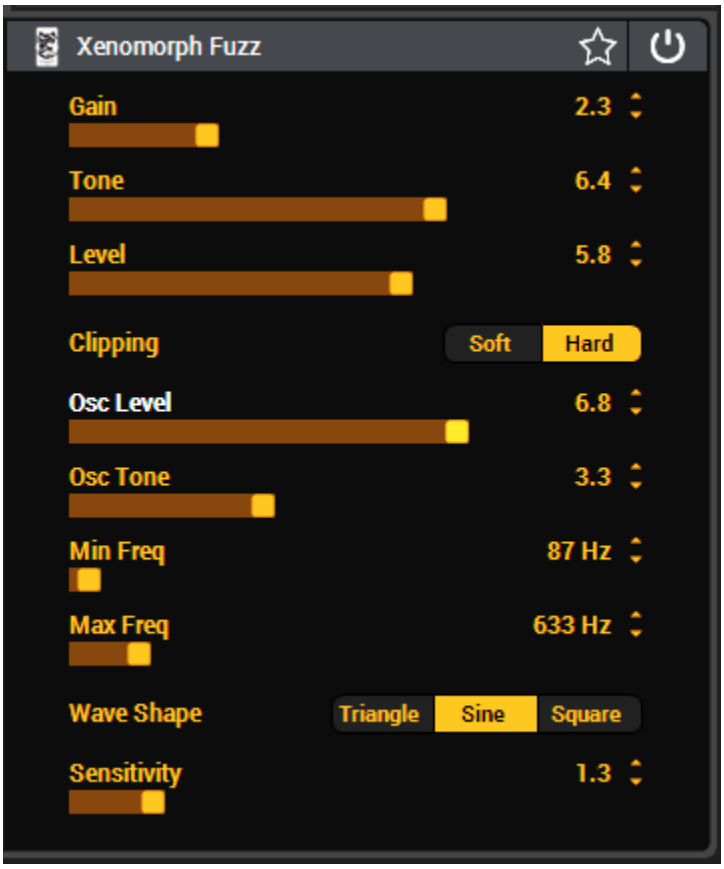


- **Drive** not only affects drive, but interacts with Attack, and changes tone. Low settings are bassier, particularly if Thick is on. High settings thin out if Attack is low.
- **Attack**'s effect is similar to a transient shaper—attacks become sharper and spikier. With Drive up full, setting Attack at maximum creates a "rounder" sound.
- **Notch Gain** cuts or boosts at the Notch Freq. Notch is a misnomer, because traditionally, notch filters only cut. Although negative parameter values cut, positive parameter values boost gain at the Notch Freq. The resulting response is like a bandpass filter.
- **Thick** gives a bassier, fatter sound when on. Otherwise, the sound is a bit thinner.

Tip: Sweeping the Notch Freq parameter with an Expression Pedal gives a wah effect that's less resonant than traditional wah + distortion combinations.

Xenomorph Fuzz

Although Xenomorph Fuzz seems intended as a distortion for chaos/noise/industrial applications, this effect is more versatile than it might appear. It's even possible to obtain sweet, square wave fuzz sounds that recall the Ultra-Fuzz project in my book *Electronic Projects for Musicians*. One difference compared to most distortion effects is that the distortion's character doesn't change much as the note decays—it just fades out.

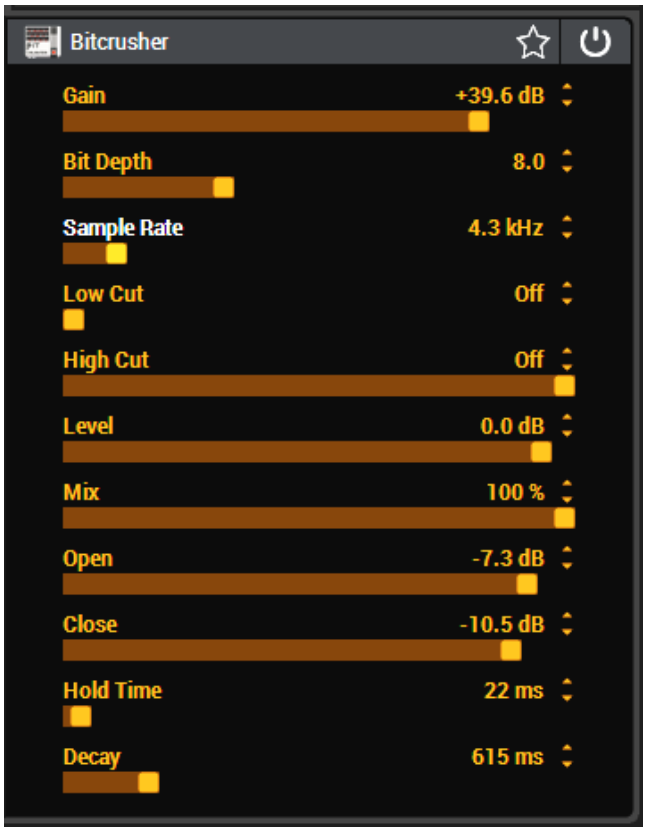


Here are some tips on using its unique features:

- For square-wave distortion, turn **Osc Level** to 0.0. **Gain**, **Tone**, and **Clipping** determine the distortion sound. I prefer the neck pickup for a sweet tone—it's somewhat like the solo in "American Woman."
- To introduce the tracking oscillator chaos, turn up **Osc Level**. The **Osc Tone** and **Wave Shape** parameters determine the oscillator's sound quality. Turn Osc Tone down for a subtler effect or turn it up for more howl.
- The oscillator frequency tracks your dynamics, as adjusted by the **Sensitivity** parameter. **Min Freq** sets the oscillator's lowest frequency. As you play harder, the frequency approaches the **Max Freq**. You don't necessarily need high Sensitivity settings to vary between these two extremes; higher sensitivity tends to kick the oscillator to the Max Freq more easily.
- To create a background "drone," set Min and Max Freq to the same frequency, based on the key you're using. For example, if you're playing in the key of A, set both to 440 Hz. Adjust the Osc Level to taste.

Bitcrusher

This Line 6 original alters the resolution of digital audio, to provide lo-fi effects. The block also incorporates Low and High Cut EQ, and a Hard Noise Gate.

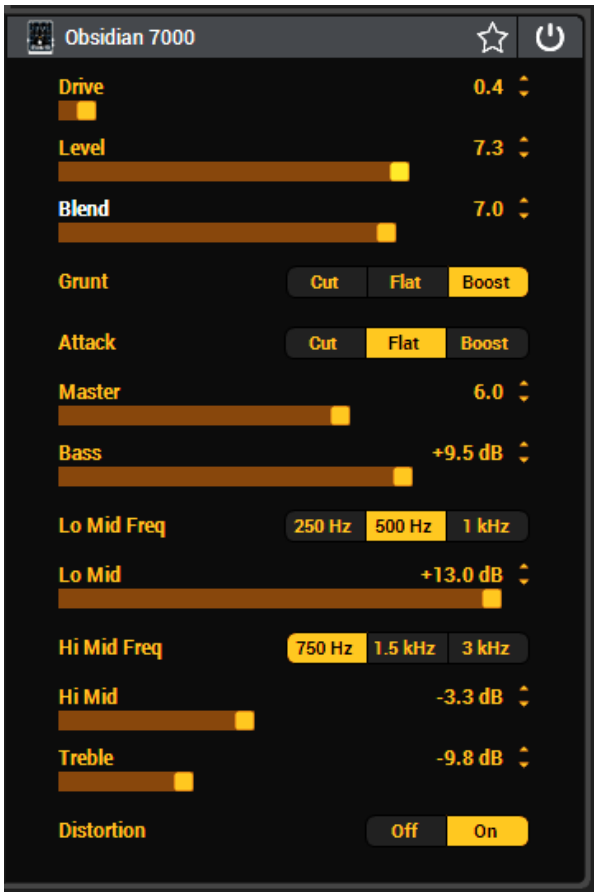


- Increasing **Gain** emphasizes the distortion, by overdriving the bit-crushing audio engine.
- **Bit Depth** (1-bit to 24-bit) alters the audio's resolution. Lower numbers give less resolution.
- **Sample Rate** (100 Hz to 48 kHz) introduces distortion when the frequency is the same as, or lower than, the frequency of incoming audio.
- **Low Cut** cuts up to 500 Hz. This is different from the standard Low and High Cut EQ block, which cuts up to 1 kHz.
- **High Cut** cuts down to 500 Hz. This is different from the standard Low and High Cut EQ block, which cuts down to 1 kHz.
- **Mix** chooses the blend of dry and bitcrushed sound.
- **Open**, **Close**, **Hold Time**, and **Decay** are the same parameters as the Hard Gate, but this gate does not include Gain Reduction metering. See Hard Gate for more information.

Tip: The lower the bit depth and sample rate, the more your guitar will sound like the bad digital audio in horrible on-hold music (or kid's toys). Those who want to take a trip down digital audio's memory lane can emulate the sound of 8-bit and 12-bit samplers, nasty converters in early CD players, the 32 kHz sampling rate of digital satellite broadcast audio, and more.

Obsidian 7000

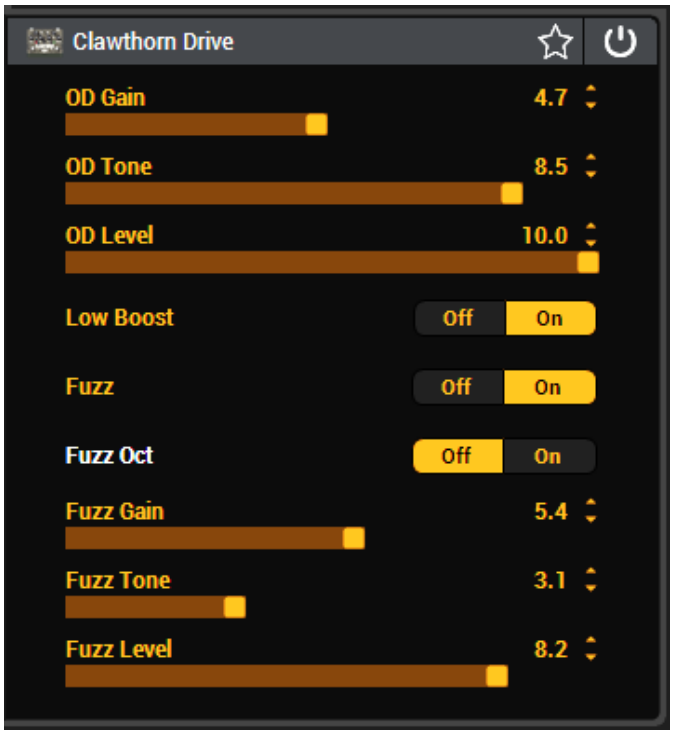
Although intended for bass, this effect also works well for big, overdriven rhythm guitar sounds. The four-band EQ is its main unique feature. It affects the clean sound, with frequencies and bandwidths optimized for bass instead of being general-purpose EQ. Because the EQ is pre-distortion, boosting a band pushes more signal through distortion in the associated frequency range, while cutting reduces distortion. This gives more control over the distortion's *character* than placing EQ after distortion, which affects only the tone.



- With **Distortion** on, **Blend** crossfades between clean and distorted tones. With Distortion Off, Blend has no effect.
- **Level** mixes the desired amount of distortion into the blend.
- **Drive** affects only the distorted signal, by altering the saturation.
- With all EQ settings flat, the clean response is flat, but the distorted response has a lift around 200 Hz and 2 kHz, with a shallow scoop in between.
- **Attack** and **Grunt** can either cut, boost, or not affect high and low frequencies respectively, prior to distortion. They don't affect the clean sound.
- **Bass** and **Treble** are shelving EQs that kick in around 500 Hz. They affect the clean sound and can also push more signal into distortion.
- **Lo Mid** and **High Mid** are parametric stages with three fixed frequencies each, and wide bandwidths that are ideal for shaping tone.

Clawthorn Drive

This is another “does more than one thing” distortion device. Its two parallel processors, Overdrive and Fuzz, complement each other.



The two sections have independent Gain and Tone parameters that work as expected, and their respective **Level** controls set the proportion of the two signal paths. Several controls have specialized functions:

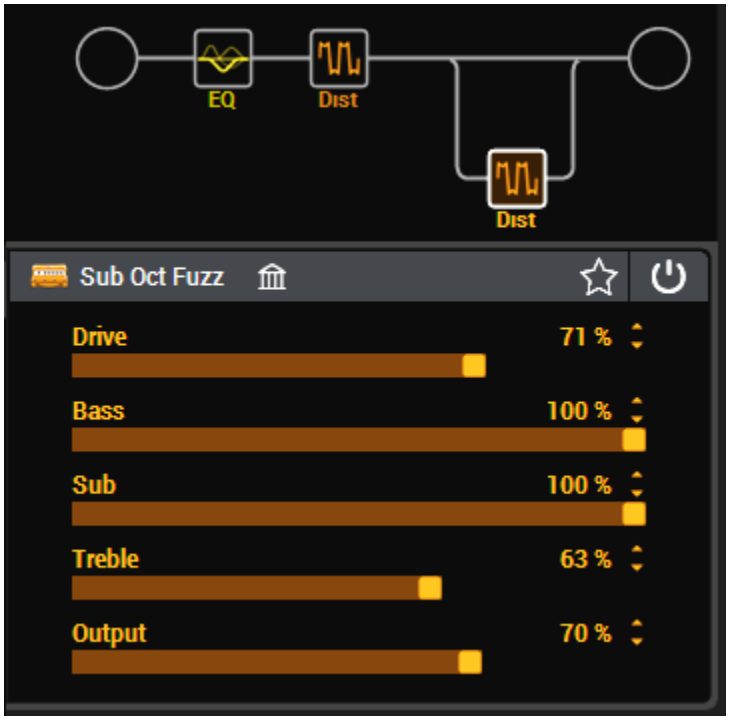
- **Low Boost** affects only the Overdrive section. On gives a beefier sound.
- **Fuzz Off/On** switches the Fuzz section in or out of the signal path. For example, you might assign this to a footswitch and use Overdrive for rhythm guitar, then turn on Fuzz for leads.
- **Fuzz Oct**, when On, produces an octave-above effect like the Tycoctavia. However, the Clawthorn Drive version sounds cleaner to me.

If you prefer a significant boost when turning on distortion, this effect has a somewhat lower maximum level than you might want. If needed, follow it with a Gain block that’s enabled by the same footswitch as the Clawthorn Drive effect.

Sub Oct Fuzz

Yes, it's a Legacy effect, and this book doesn't really cover the Legacy models. But I designed the PAiA Roctave Divider—the original hardware circuit on which this was based—so I figure I'm entitled to comment. Line 6 got close, but they left out an important control: in addition to the octave down knob, the hardware version had *another* control for two octaves down.

Hardware Roctave Dividers are an endangered species. I've seen them go for as much as $300 used. But that's not the only way to get the official extra-octave-lower-sound! Hey, if people can mod my stuff, I can mod Line 6's…right?



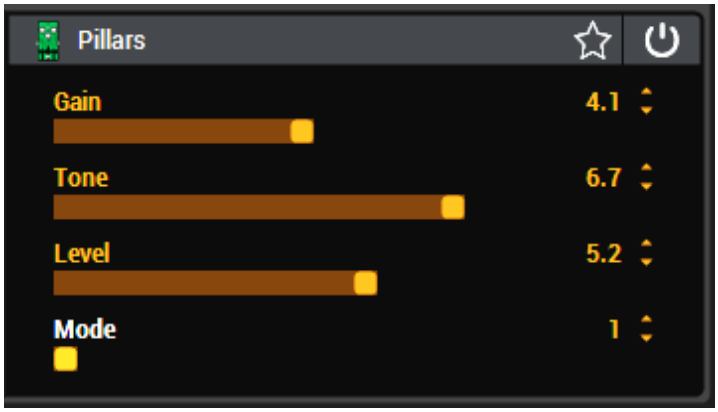
The **SubOctFuzz_CA** preset is in the *Free Files > Chapter 2 Presets* folder. The modification is simple. The first Sub Oct Fuzz that provides the one-octave-down sound feeds into a paralleled Sub Oct Fuzz that's also set for an octave down. So, the second Sub Oct Fuzz divides the octave down signal by *another* octave, and there you have it—the missing two-octaves-down control. Mix the two paths in the Mixer.

Another mod is that the Low and High Cut filter at the beginning improves the tracking.

The most recent description I saw from someone selling a Roctave Divider online said it is "one of the loudest and meanest octafuzz out there." I consider that a compliment…and now you can obtain the missing ÷4 sub-*sub*-octave control in Helix.

Pillars OD

This draws its inspiration from the Tube Screamer TS808 but is definitely not an emulation. As you'd expect, Gain controls the amount of drive. Level handles the output volume.



- **Tone** isn't a simple treble or bass boost. It shifts the boosted frequencies around the midrange. Although it leaves the lower frequencies mostly unaffected, the high-frequency response changes at different settings (fig. 2.5).

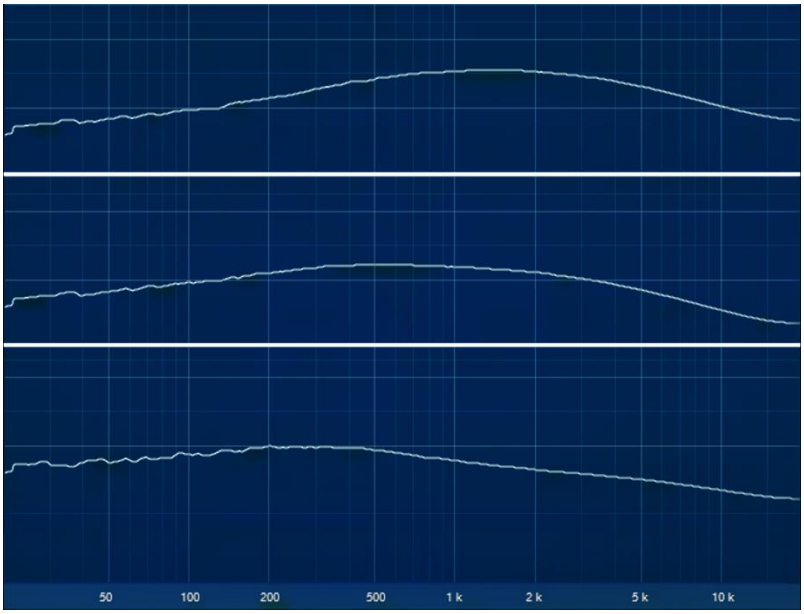


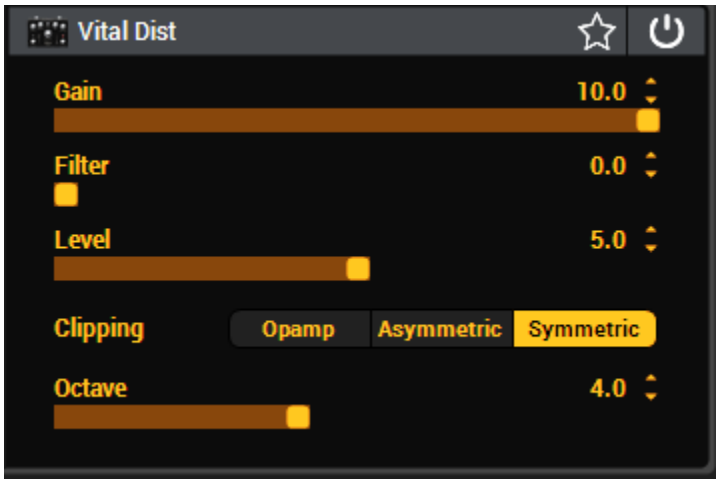
Figure 2.5 Tone control with mode 1. Lowest image is Tone = 0.0, middle is Tone = 5.0, top is Tone = 10.0. The tone shifts from a bass emphasis around 300 Hz to a treble emphasis around 1.5 kHz. Even though the 10.0 setting boosts the highs compared to the other settings, the midrange remains prominent.

- **Mode** differentiates Pillars further from the TS808:

    - Mode 1 uses LED-based, symmetrical clipping. Its distortion character is bright and crunchy.
    - Mode 2 is wide-open op amp gain. This produces square wave-like distortion. It also creates a much higher output. Before switching into Mode 2, consider trimming the Level.
    - Mode 3 is a variation on Mode 1. Its silicon diode-based distortion uses asymmetrical clipping. The sound is like a kinder, gentler Mode 1. There's less brightness and crunch.

Vita Dist

Vita Dist emulates the Amplitude section of Earthquaker's Life pedal. This supplements a three-mode distortion pedal with an octave up control. The three Clipping modes resemble the ones for Pillars (see the previous description). However, Mode 2 models two silicon diodes with an LED, and Mode 3 models back-to-back silicon diodes. Vita Dist has a more aggressive, "spikier" sound than Pillars. It's like the difference between fuzz/distortion and overdrive, respectively.



The Filter control is basic. Turning it up higher reduces the high frequencies (fig. 2.6), so you can tame the degree of brightness. This also reduces the presence caused by crunch.

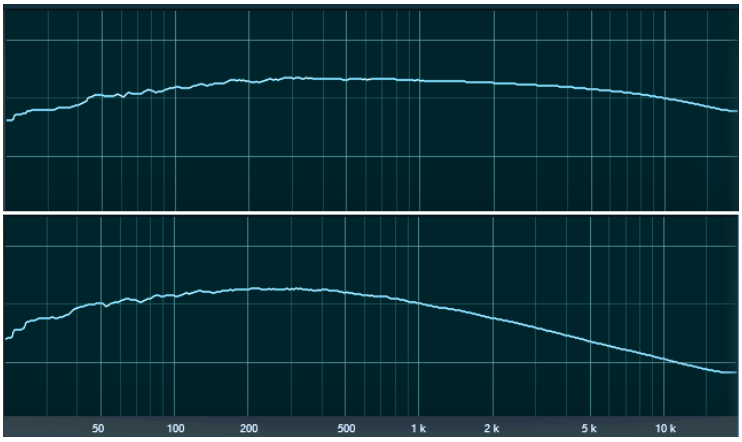
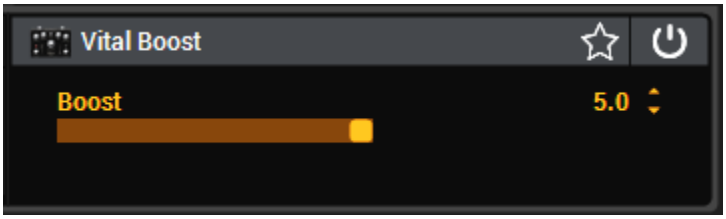


Figure 2.6 The upper image shows Filter at 0.0, the lower image has Filter at 10.0. The amount of high-frequency attenuation doesn't change much with the different clipping options.

The Octave control adds an octave-above waveform. Like other octave multipliers based on full-wave rectifier circuits, it works best with single-note parts, like leads.

Vita Boost

Vita Boost may sound like something you'd find in a health food store, but it also relates to Earthquaker's Life pedal. Life isn't just about the Amplitude section, but an included Boost (the Magnitude section). If you don't like lots of controls, this is the droid for you.



Boost's goal is simply to produce gain. It adds no distortion of its own. However, you *can* overdrive it so that a sine wave turns into a square wave with overshoot. Interestingly, between no distortion and the square wave, there's a subtle, asymmetrical rounding off that's quite tasty (fig. 2.7). Boost doesn't affect this, so if you find a "sweet spot" overdrive level, you can use the Boost parameter to set the overall level.

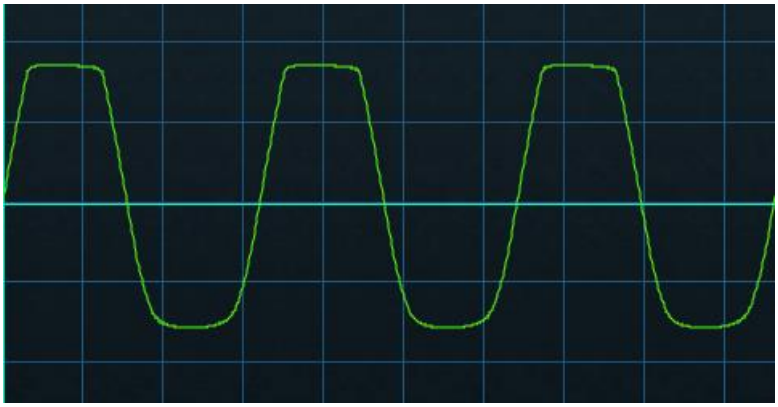
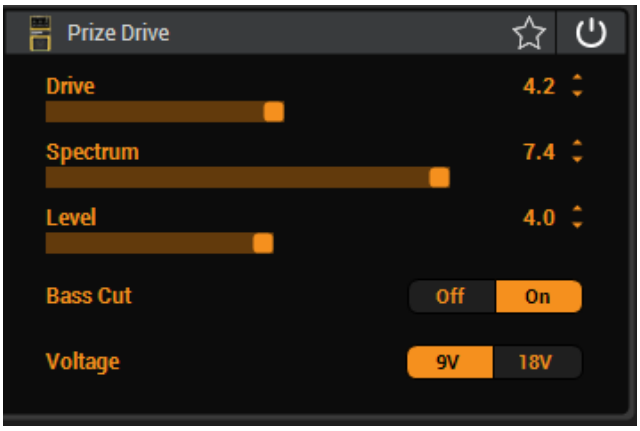


Figure 2.7 The onset of distortion in Boost has an interesting, sweet character, as long as you don't go too far.

To produce a full simulation of the Life pedal, follow the Vita Dist by the Vita Boost. This is the same order of effects as in Life. Like Life, you can bypass them independently. This is helpful for when you have a Vita Dist sound you like for rhythm, but then want a massive boost for additional rhythm crunch, or lead parts.

Prize Drive

This is based on a pedal that's a staple in the Nashville scene. It's not quite like other overdrive pedals. It arguably gives more of a modern country/rock sound than a swampy blues crunch.



- **Drive** covers a wider range than you might expect. Full up gives heavy distortion.
- **Level** is an output control. Turn this up to overdrive your amp.
- **Bass Cut** thins the low end, which some guitarists find too prominent.
- **Voltage** can select 9V, or 18V for more headroom if needed.
- The **Spectrum** control (fig. 2.8) is unique.

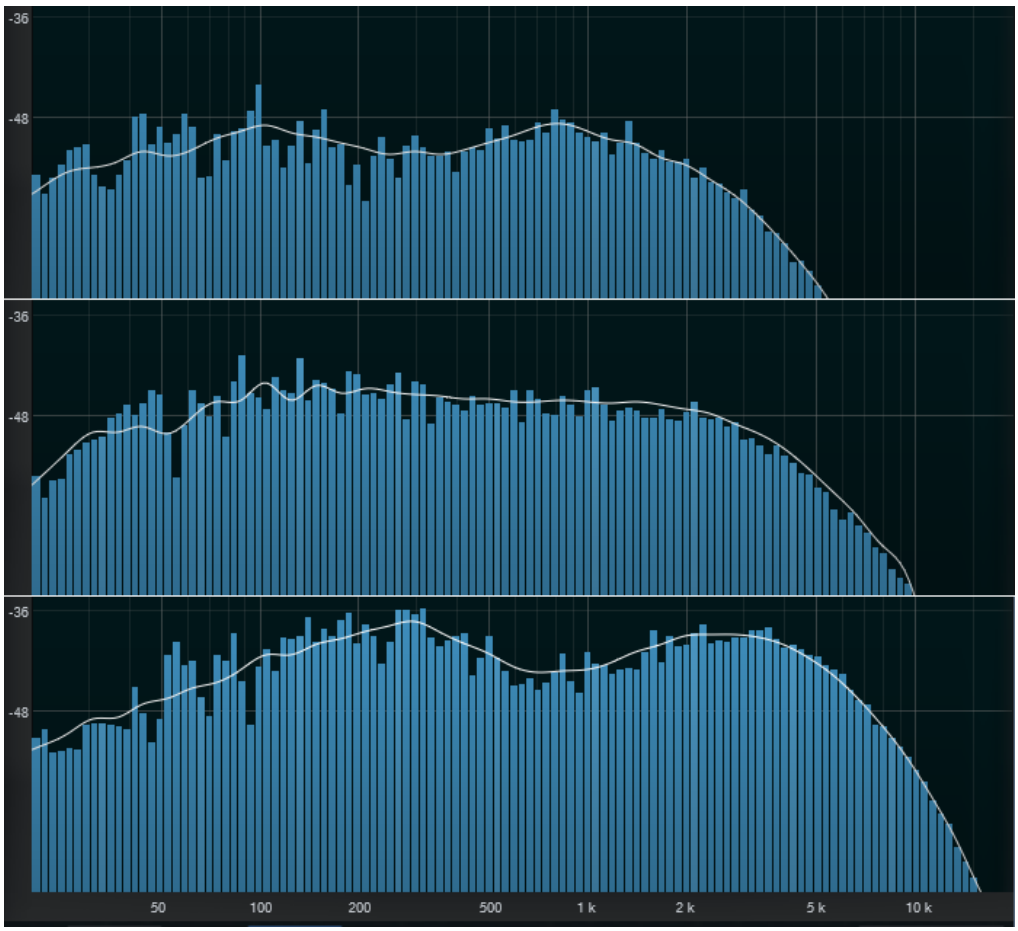


Figure 2.8 The Spectrum control isn't just about high- or low-frequency response.

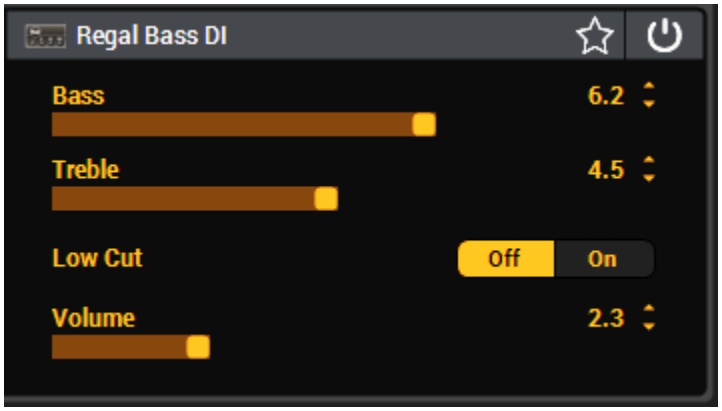
In fig. 2.8, the top image shows the Spectrum control at 0. Note the scoop around 300 Hz, the lifts around 100 and 800 Hz, and the attenuated high-frequency response.

The middle image is the Spectrum control at 5.0. The response is flatter, with a bit more low end, and the highs starting to increase.

The bottom image shows Spectrum up all the way to 10.0. Now there's a big boost around 300 Hz (ideal for giving single-coil pickups some humbucker-like "beef"), a significant midrange scoop around 800 Hz, and a lot more highs. The highs max out around 3 kHz, but the high-frequency response is generally more extended.

Regal Bass DI

This goes into the “small but mighty” category. I consider this more as a “DI with color” than a typical distortion device. The parameters are simple: Bass and Treble controls, Low Cut switch, and Volume.



The emulated tube path adds musical harmonics to bass, as underscored by fig. 2.9. A 100 Hz sine wave has no harmonics of its own, so the Regal Bass DI is what adds the harmonics in the screenshot. The strongest harmonics are at 200 Hz (+1 octave), 300 Hz (octave above+fifth), and 400 Hz (+2 octaves). The higher harmonics are weaker. This type of harmonic distortion is characteristic of tubes, but not solid-state devices. This is one reason why many musicians feel tubes have a “sweeter” sound.

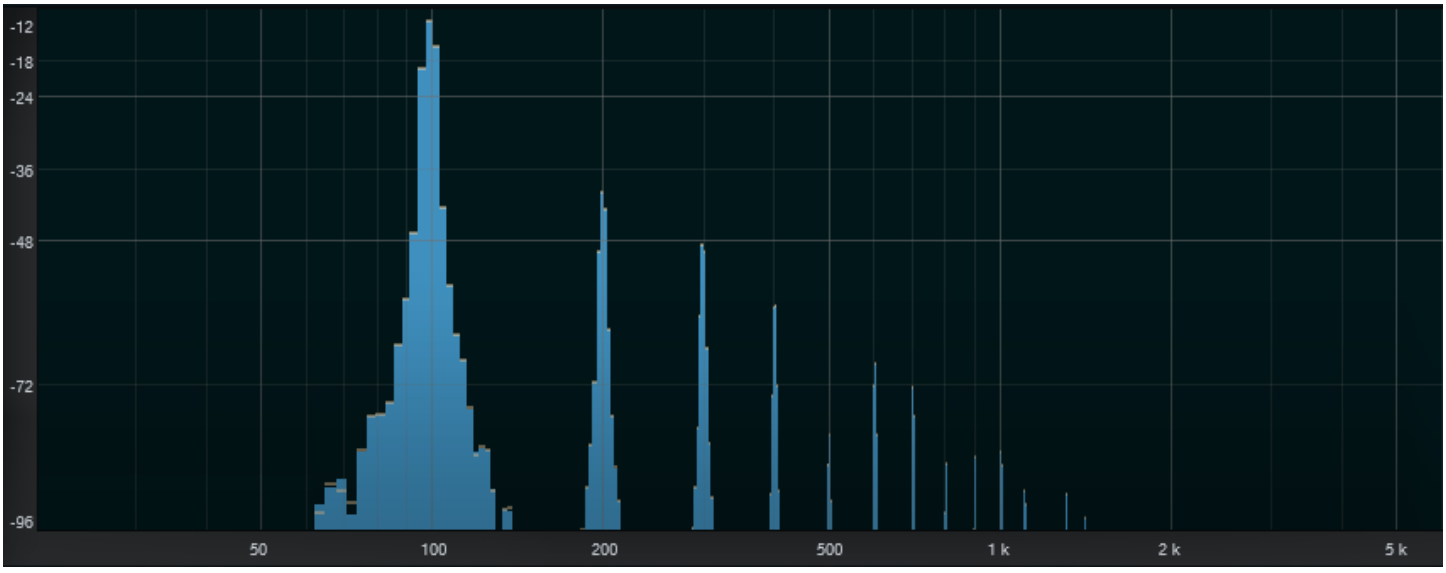


Figure 2.9 Harmonic distortion for a 100 Hz sine wave.

Regarding the tone parameters, the response is essentially flat when both parameters are at 0.0. In fig. 2.10:

- The upper image shows the response with Treble at 10.0 and Bass at 0.0.
- The middle image shows the response with Bass at 10.0 and Treble at 0.0.
- The bottom image shows the Bass and Treble at 0.0, but with the Low Cut enabled.

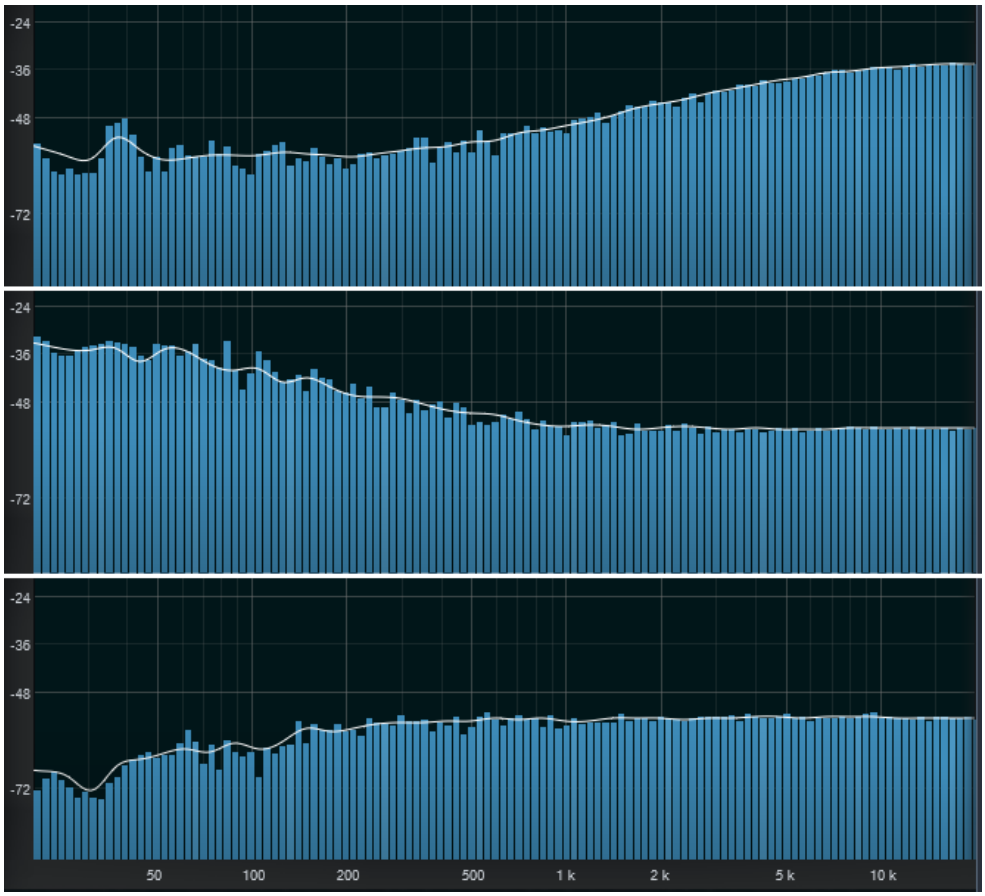


Figure 2.10 Tone control frequency response curves.

Don’t let the name (or the simplicity) throw you off! This is a wonderful guitar processor that can make a clean guitar really “speak.” Indeed, the Regal Bass DI is small but mighty.

Dynamics Blocks

Dynamics processors alter level. The sustainer, based on compression circuitry, is the most common dynamics application for guitar. A sustainer is simpler to adjust than a studio-oriented compressor because it has only one purpose, so it requires fewer controls. In addition to compressors, Helix includes two other dynamics processor types: the limiter (in the LA Studio Comp section), and noise gate.

About Compressors

Compressors are some of the most used—and misunderstood—signal processors. People often try compression to make a sound "punchier," but they instead dull the sound because of improper control settings.

A compressor evens out dynamic range variations, by reducing the difference between soft and loud levels. Fig. 2.11 shows how different compressor parameter settings create different guitar sustain characteristics.

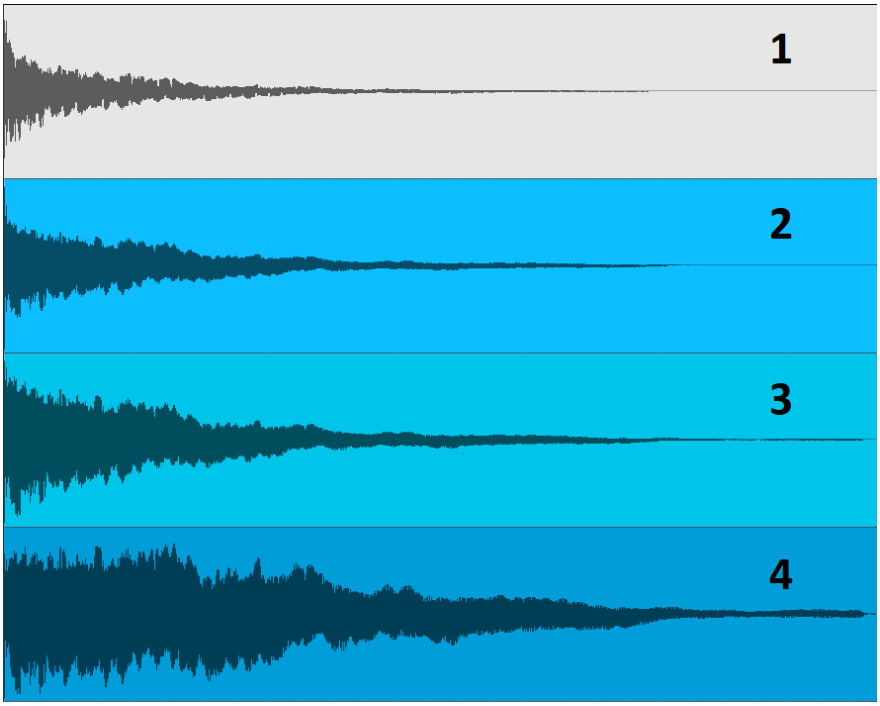


Figure 2.11 (1) is an uncompressed, decaying guitar chord. (2) shows moderate compression on the sustain but leaves the initial attack intact. (3) uses the same settings but compresses the entire signal, including the attack. This allows raising the overall level considerably. (4) uses extreme amounts of compression to act as a sustainer.

Lowering a signal's peaks opens up more headroom, so you can raise the overall level. For example, suppose some peaks reach 0 dB, and compression reduces those by -5 dB. The loudest peaks now reach -5 dB. So, you can turn up the level by +5 dB, and the peaks once again hit 0. By reducing the peaks, you're able to add +5 dB of gain to the overall signal, and make its average level louder,.

Studio-oriented compressors are more difficult to adjust than sustainers not only because they have more controls, but because the controls interact. The tradeoff is being suitable for more types of applications:

- Increase touch sensitivity.
- Reduce level peaks that cause distortion with downstream effects.

- Smooth the sound of distortion effects and high-gain amps.
- Maintain a consistent level, so guitar doesn't get "buried" in a mix.
- Make some effects, like flangers and phasers, more dramatic.

Tip: Don't overlook Helix's Legacy compressors. How people react to compressors is subjective, and one of these might be what you want.

Signal Chain Placement

Where you place the compressor in the signal chain matters:

- Inserting a compressor after noisy effects brings up the noise, so it's not recommended. A limiter may be a better choice for simply controlling peaks further down the effects chain.
- Compression before distortion or high-gain amps increases sustain. However, remember that distortion also compresses the dynamic range by cutting off peaks. Too much compression before distortion can give an unfocused sound, with less dynamic range.
- An envelope-controlled effect (e.g., the Autofilter) may be too sensitive to your dynamics—picking hard can push the filter frequency too high, while playing softly pulls it down too far. You can compensate for this by tweaking the effect's controls, but the response might still be touchy. Inserting the LA Studio Comp before the filter can restrict the dynamic range enough to give a more predictable response.
- To tame a resonant filter or flanger's peaks, insert compression after the processors to make the dynamics more consistent.
- Compression interacts with EQ. EQ *after* compressor changes the compressed sound's tone. Boosting EQ *before* a compressor compresses the boosted frequencies more easily than other frequencies. For example, boosting around 3 kHz before compression makes solos at the top of the neck more sensitive to your touch.

  Fig. 2.12 shows the same riff, with each riff played one octave higher than the one to its left. The lower image, with compression and an EQ boost at 3 kHz, brings up the riff's level with higher notes.

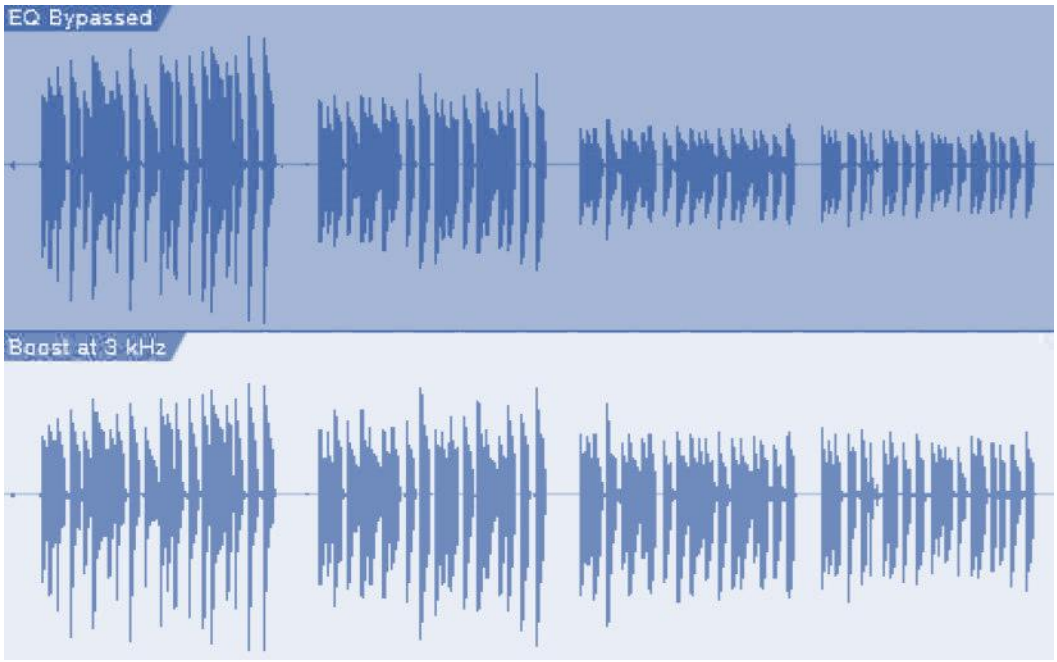
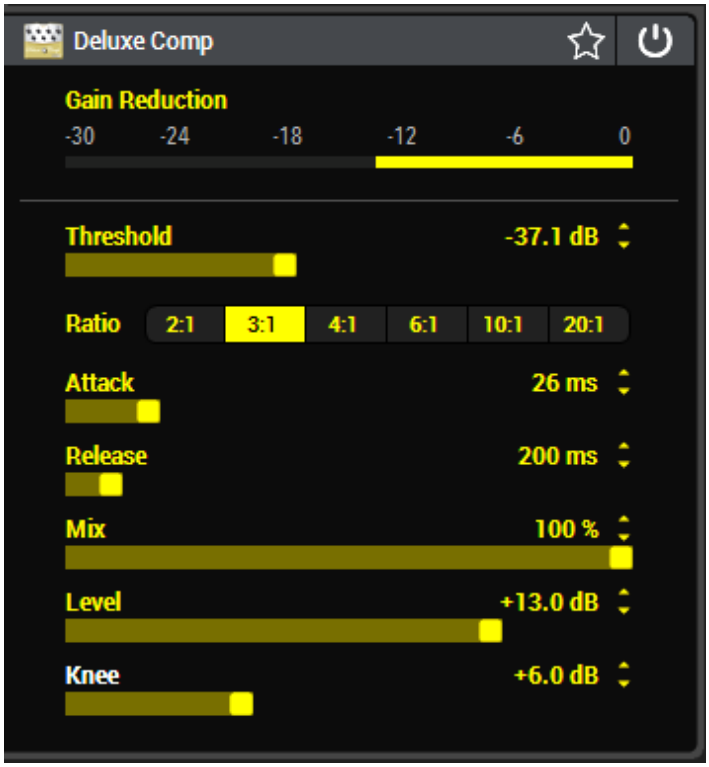


Figure 2.12 Placing EQ before compression gives a more even, sustained sound with solos played on the higher frets.

Deluxe Comp

The Deluxe Comp strikes a good balance between the radical sustain characteristics of the Red Squeeze, and the more transparent vibe of the LA Studio Comp. However, it's more challenging to dial in the optimum settings with a multi-parameter, complex processor like the Deluxe Comp. Its parameters are like those used in recording studio-type compressors, and it's well-suited to different compressor applications.



Understanding what the parameters do is vital to getting the best sound:

- Above the **Threshold**, the output level increases by a lesser amount than the input level. So, lower thresholds compress more of the signal.
- **Ratio** is the ratio of output-to-input-level increase. For example, with a 4:1 ratio, when the input signal is above the threshold and increases by 4 dB, the output increases by only 1 dB. Higher ratios make the compression effect less transparent.
- The **Gain Reduction** meter (Helix Native only, not in HX Edit or other Helix versions) shows how much the compressor is reducing gain to meet the requirements for output signal level vs. input signal level, as specified by the Threshold and Ratio parameters.
- **Attack** determines the time it takes for the compressor to react fully to an input level change. Longer attack times let through more of a signal's initial dynamics.
- **Release** sets the time required for the compressor to return to no compression once the input passes below the threshold. Shorter settings track the input signal's level changes more closely. Longer settings cause more gradual level changes.
- **Mix** sets the ratio of dry to compressed sound. Including some dry signal preserves some of the signal's natural dynamics. This technique is common with drums.
- **Level** is almost always greater than 0, to compensate for the loss in output level caused by compression. Generally, adjust Level so that whether the compressor is bypassed or enabled, the guitar's *peaks* are the same. Enabling the compressor then gives a subjectively louder, more "in your face" sound.

- **Knee** controls how rapidly the compression kicks in. At 0.0, when the input signal crosses the threshold, it's immediately subject to the full amount of compression. Increasing the knee makes the compression action less abrupt. Light compression begins below the Threshold. It increases to full compression above the Threshold.

Getting the Right Sound

Understanding how the parameters interact helps you get the sound you want:

- The input level has a major effect on compressors. If the Threshold is -10.0 dB and your guitar's peaks never exceed -10 dB, you won't hear any compression effects. If the guitar's peaks are around -2 dB, much of the audio will exceed the threshold, and you'll hear a fair amount of compression.
- In the studio, I've optimized my compressor presets for a nominal peak input level of -3 dB. Then if a track is going to feed the compressor, I edit the track's level so that its peaks hit -3 dB. This gives a predictable result from presets that are optimized for this input level.
- Threshold determines the range of levels that will be compressed, while Ratio applies the amount of compression to that range of levels. If the compressed guitar sounds unnatural or "squeezed," raise the Threshold to compress less of the signal or lower the Ratio so that the compression has less effect.
- Try choosing a low Ratio at first. Then, lower the Threshold to obtain the desired compressed sound. If it's not compressed enough, choose a higher ratio, then re-adjust the Threshold.
- The most natural effects occur with low compression ratios. 2:1 to 4:1 ratios are common in the studio for adding a touch of compression to acoustic and electric guitar. 6:1 and 8:1 give more of a compressed effect. 10:1 visits the land of sustainers, and 20:1 books a room there.
- Bass often uses lots of compression. This is because playback systems have difficulty reproducing low frequencies. So, a consistent bass level has a better chance of being heard.

Red Squeeze, Kinky Comp, Rochester Comp

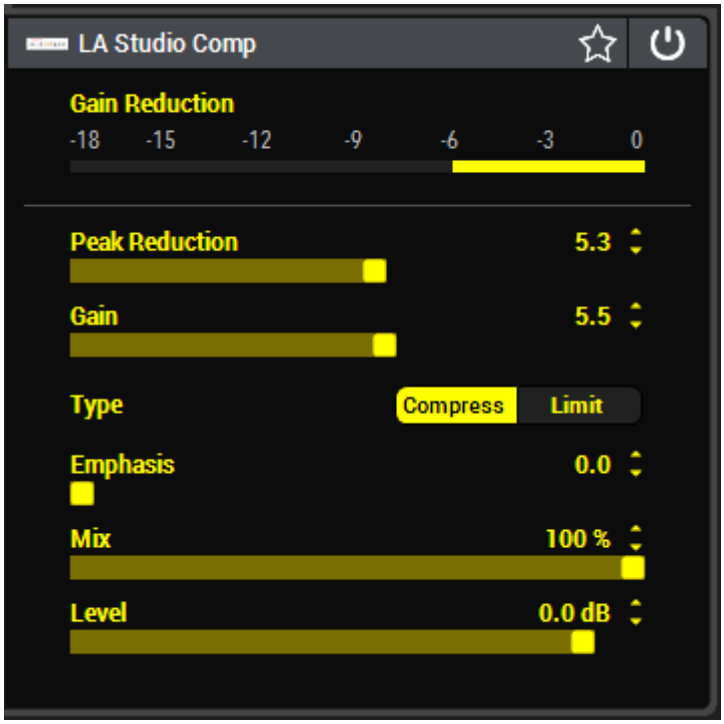
The parameters in these three compressors resemble those in the Deluxe Comp but produce different results. Note that the **Gain Reduction** meter is in Helix Native only, not in HX Edit or other Helix versions.

Here are some opinions:

- **Red Squeeze** has the most "stompbox sustainer" character, with a savage attack. The **Sensitivity** parameter replaces the traditional Threshold and Ratio. Setting the **Mix** control to an approximately equal mix of dry and compressed sound gives a beefier compression effect than the Deluxe Comp, but without the extreme compression effect of the "native" Red Squeeze sound. Note that it's not possible to obtain unity gain, because the output level will always be lower. Insert the Red Squeeze before amps for more sustain, or to avoid slamming the amp so hard with level that you lose definition.
- **Kinky Comp** is a gentler version of a stompbox sustainer, where **Sensitivity** sets the amount of compression. Compared to the Red Squeeze, it has a cleaner, less aggressive character that works well when placed before other effects (or even after effects, in some cases).
- **Rochester Comp** is clean and transparent. Technically speaking it's a *leveling amplifier*, which combines compression and limiting to maintain as consistent an output level as possible. It's ideal for compressing nuanced guitar sounds, like ones that incorporate modulation. It also works well in the studio for program material. It doesn't give that "punch-your-guitar-attack-in-the-face" stompbox compressor effect, but Helix has other options for that (like the Red Squeeze).

LA Studio Comp

This isn't a stompbox-type sustainer, but models an iconic, high-end studio compressor that has been used for decades on vocals, bass, guitar, and more. Helix Native users might find themselves inserting this effect into a variety of tracks.



- The **Gain Reduction** meter (Helix Native only, not in HX Edit or other Helix versions) shows how much the level is being reduced to obtain the specified amount of peak reduction.
- **Peak Reduction** determines the amount of gain reduction. Higher values compress more.
- **Gain** makes up for the loss of output level caused by Peak Reduction.
- Turning up **Emphasis** compresses high frequencies more than lower frequencies. A maximum setting of 10.0 adds a slight sparkle. It also causes less "pumping" or "breathing" on lower-frequency signals. Turning it down to 0.0 gives an even compression effect over all frequencies.
- **Mix** allows for parallel compression, which blends the dry and compressed sounds. This combines the compressed sound with a dry sound that still maintains some dynamics.
- It's often best to keep **Level** at 0.0 and adjust the Gain control for unity gain (if that's what you want).

Type (Compress and Limit)

Compressors and limiters have the same goal of reducing dynamic range, but do so differently:

- **Compress** acts like a standard compressor—above a certain threshold, the output level increases less than the input level.
- **Limit** clamps the output to a fixed level, so that no matter how much you turn up the input, the output stays constant. Think of it as a compressor with a 100:1 compression ratio instead of the usual 2:1 to 10:1 range.

Limit applications include:

- Insert the LA Studio Comp at the *end* of a chain so that the Helix output doesn't exceed a certain level.
- With slap bass, the transients can be super-high. Insert limiting early in the chain to keep these from getting out of hand.
- When recording, use limiting at the output so peaks don't overload the input, and cause distortion.
- Inserting limiting before a sustainer can help reduce "popping" at the beginning of notes.

Tech Talk: What Makes the LA Studio Comp Special

The compression technology used in the original LA-2A (and LA-3A) is based on the T4 opto-isolator, which combines a photo-resistor and electro-luminescent panel. Audio feeding the panel changes the amount of light that shines on the photo-resistor, which alters the level and therefore, the gain. The opto-isolator's inherent response is a "happy accident" for audio—it's almost as if it comes with a built-in auto-attack/release switch.

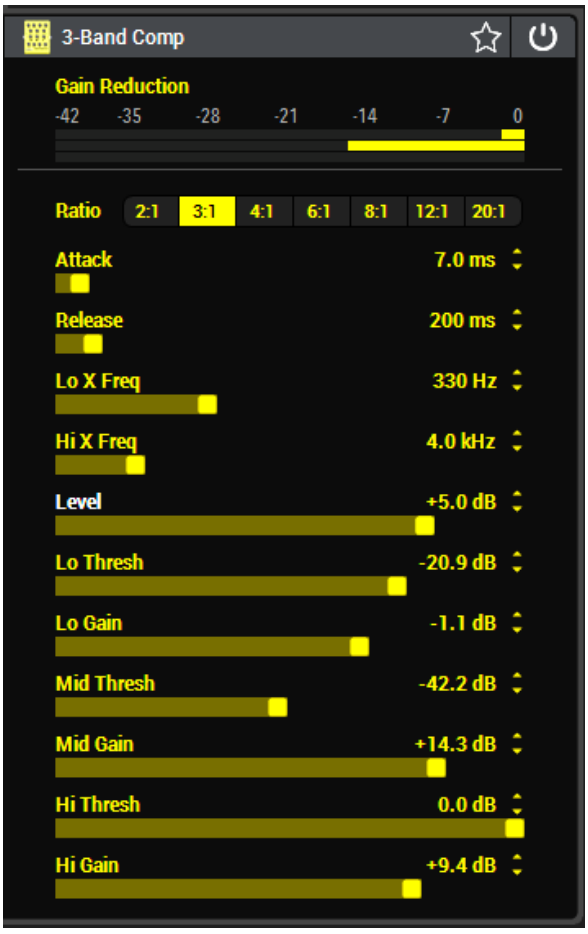
The fixed, non-linear attack doesn't remove all the punch, and the release occurs in two stages, where the release time decelerates over time. (Even some modern hardware designs that use LEDs instead of electro-luminescent panels have a similar response.) Note that a major advantage of a virtual LA-2A compared to the hardware is that a virtual electro-luminescent panel doesn't deteriorate over time.

With vocals, the attack time retains the consonants at the beginning of words, for greater intelligibility. With guitar and bass, it lets through the initial punch of hitting a string, but then settles down to compression with a smooth character.

There are also some fine points. For example, with strong signals the release time lasts longer. But the release time may not be complete before the compressor gets hit again with signal, resulting in an attack time that's shorter than expected. Conversely, if the release does play out all the way, then there will be the full attack time when the next audio hits. This is mostly of academic interest—in the studio, "it just works."

3-Band Comp

This is one of my favorite processors, because it can do much more than compression…as we'll find out later, in the 3 and 4-Band Processing chapter. For now, we'll focus on compression.



I've re-drawn the UI to clarify the signal flow.

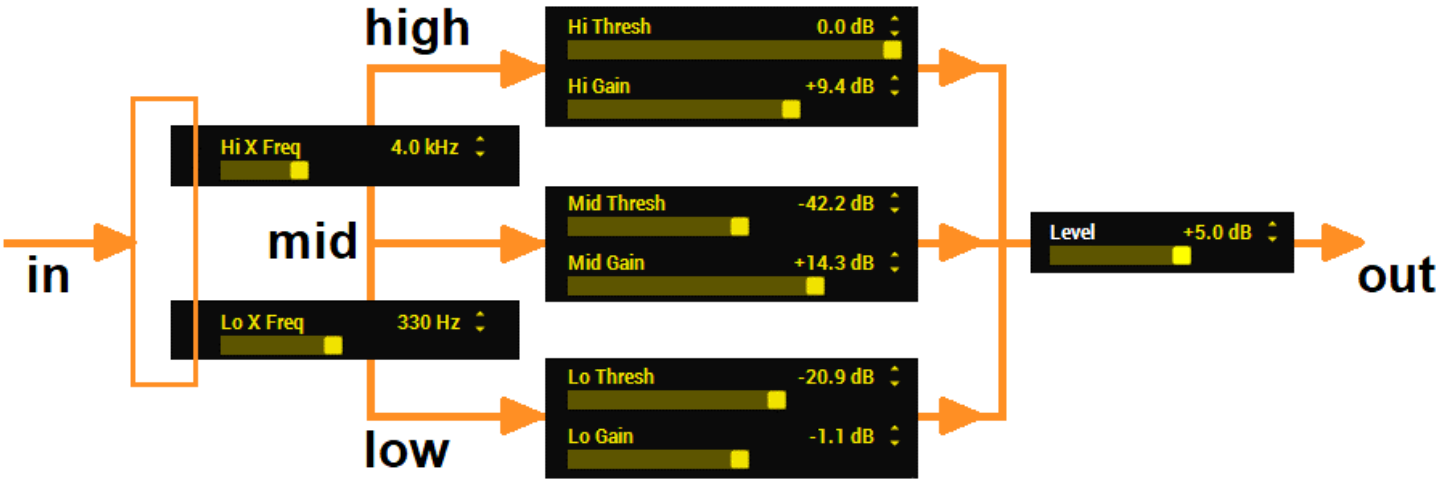


Figure 2.13 How the 3-Band Comp processes the input.

Referring to fig. 2.13, the input feeds into a crossover section that specifies two frequencies, **Hi X Freq** and **Lo X Freq**. These split the audio into low, mid, and high bands. Each band goes through its own compressor. In this example, audio above 4.0 kHz goes to the Hi compressor, audio between 330 Hz and 4.0 kHz goes to the Mid compressor, and audio below 330 Hz goes to the Lo compressor.

All the bands share common **Ratio**, **Attack**, and **Release** parameters. Note that with Helix Native (not in other Helix versions or HX Stomp), each band has its own **Gain Reduction** meter.

The Advantage of Multiple Bands

A classic example is compressing drums. With a standard compressor, a kick that causes compression will also apply compression to everything else. With a multiband compressor, you can compress the low-frequency kick, high-frequency cymbals, and midrange drums (like snares and toms) independently.

A multiband compressor also has some of the attributes of EQ. Each band has a level control, so it can act like a 3-band "targeting" graphic equalizer.

With guitar, the settings shown in the screenshot give humbucker pickups more spank:

- The Lo band has moderate compression, and not a lot of makeup gain. So, the low end is tight, controlled, and not as prominent as standard humbucker pickups.
- The Mid band has a lot of compression and gain, to bring the guitar's midrange forward.
- The Hi has no compression, but a fair amount of gain to give a bright, more single-coil kind of sound. In this case, the Hi section acts like EQ, not compression.

The 3-Band Comp is also great with bass. Do heavy compression on the low end for a full, round sound, a bit of midrange compression for articulation, and a high-end boost to bring up pick noise and string zing, to help cut through a mix.

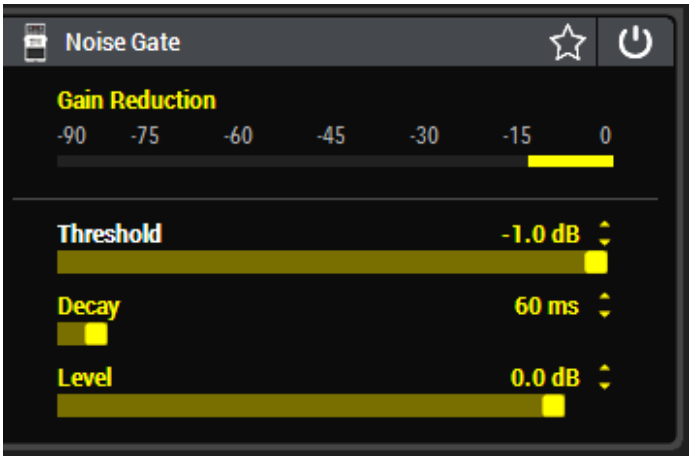
Noise Gate

Technically this is not a noise gate, but a dynamic range expander (the opposite of a compressor):

- When the input is *above* the **Threshold**, the output is the same level.
- When the input level is the *same* as the threshold, the output is slightly lower than the input.
- As the input goes further *below* the threshold, the output level becomes lower than the input (e.g., if the input is -10 dB below the threshold, the output level will be about -20 dB below the input level). This is a gentler kind of level reduction compared to the hard cutoff of a traditional Hard Gate.

Below a certain input level, the gate shuts down, or at least the output becomes inaudible. For example, with a -15 dB threshold, an input signal below -40 shuts off the "gate." The **Decay** parameter sets the time for the gate to react after audio goes below the Threshold.

Note that the **Gain Reduction** meter is only in Helix Native, not in HX Edit or other Helix versions.

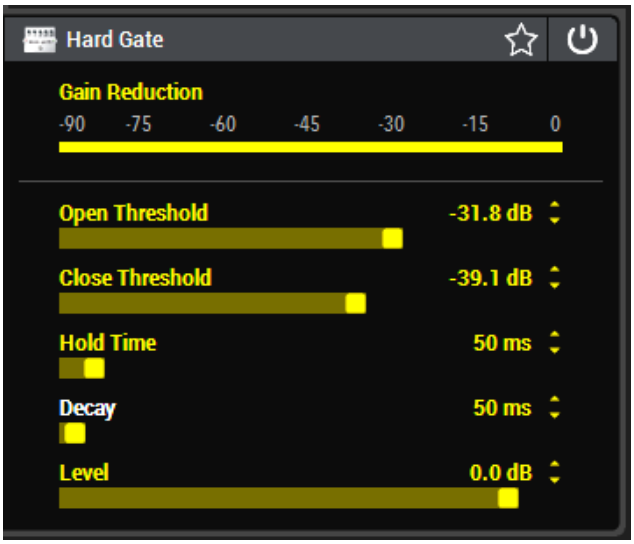


There are two main applications:

- With low Threshold settings, the Noise Gate fades into silence instead of cutting off the audio. It's less jarring than a conventional noise gate.
- High Thresholds can emphasize only the peaks of your playing. This is useful with parallel processing, as shown in the preset **Dynamic Brighten**.

Hard Gate

The Hard Gate passes audio if its level is above the Open Threshold. If the input level drops below the Close Threshold, the gate mutes the audio. In addition to suppressing low-level noise like hiss and hum, the Hard Gate can also provide creative effects. For example, it can make audio more percussive by passing only peaks.



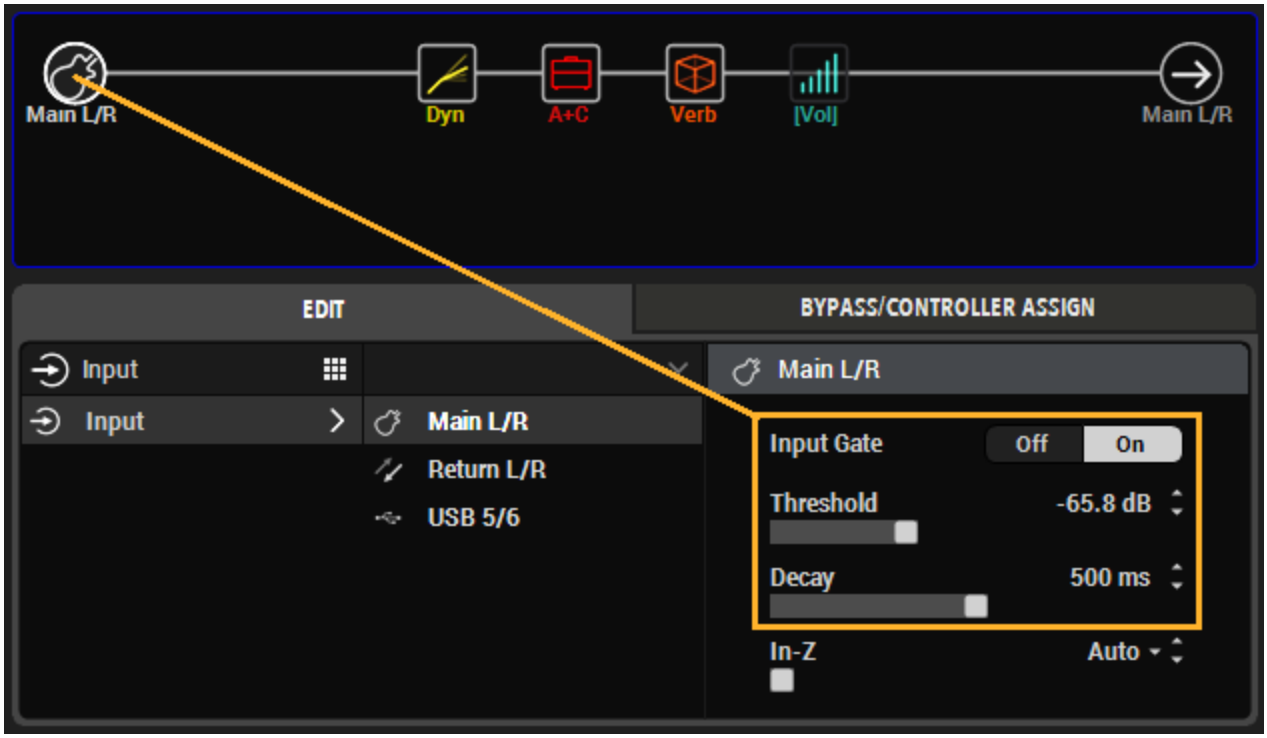
The lower the threshold, the less likely that the gating action will impact the audio that you *do* want to hear. So, when reducing noise or hiss, the Hard Gate is most effective on signals without much noise. The parameter settings are somewhat crucial:

- **Open Threshold** sets the level above which the gate lets audio through. To avoid cutting off low-level sounds you do want to hear, set the Open Threshold just above the level of any noise to be removed.
- **Close Threshold** sets the level below which the gate closes, which mutes the audio. This is usually lower than the Open Threshold.
- **Hold** keeps the gate open for the specified amount of time, after the audio goes above the Open Threshold. For example, suppose the Hold time is 100 milliseconds. If 70 milliseconds after the gate opens the level goes back momentarily below the Close Threshold, the gate will still stay open for another 30 milliseconds. If the signal goes back above the Open Threshold, the Hold time prevented the gate from cutting out momentarily. A hold time is useful for signals that cross over the threshold several times when they decay, like a decaying guitar string. Without hold, there would be a “chattering” or stuttering effect. Also, with low-frequency bass signals, at some settings individual audio cycles may turn the gate on and off. This produces a buzzing sound. Adding a few dozen milliseconds of hold time prevents this.
- **Decay** ramps the audio down from the gate-open state to the gate-closed state. This provides a smooth transition as the gate closes, instead of an instant volume change from on to off. A typical decay time is 100 to 200 ms.
- The **Gain Reduction** meter shows when the signal is being muted. Note that this is in Helix Native only, not in HX Edit or other Helix versions.

Tip: Higher Open and Close Thresholds can create super-percussive guitar sounds by letting through only the peaks. Try this on stabs, and quick downstrokes or upstrokes.

Input Gate

The Input block is almost identical to the Noise Gate. The input is a logical place to have a gate, because it can remove low-level hum and noise from the pickups, before it gets into other effects.



Like the Noise Gate, the **Threshold** is a single parameter, as opposed to the Hard Gate's separate Open Threshold and Close Threshold parameters. **Decay** works the same as the Noise Gate. Because there's no hold parameter, adding some decay can help smooth out any chattering that occurs if the decaying string crosses back and forth over the threshold.

For many situations, this gate will do what you need, so you don't need to "spend" a block on a dedicated gate.

Horizon Gate

This is not a traditional noise gate, but a noise *filter*. As the input signal gets lower, there's a greater amount of high-frequency rolloff (fig. 2.14). This is ideal for reducing hiss.

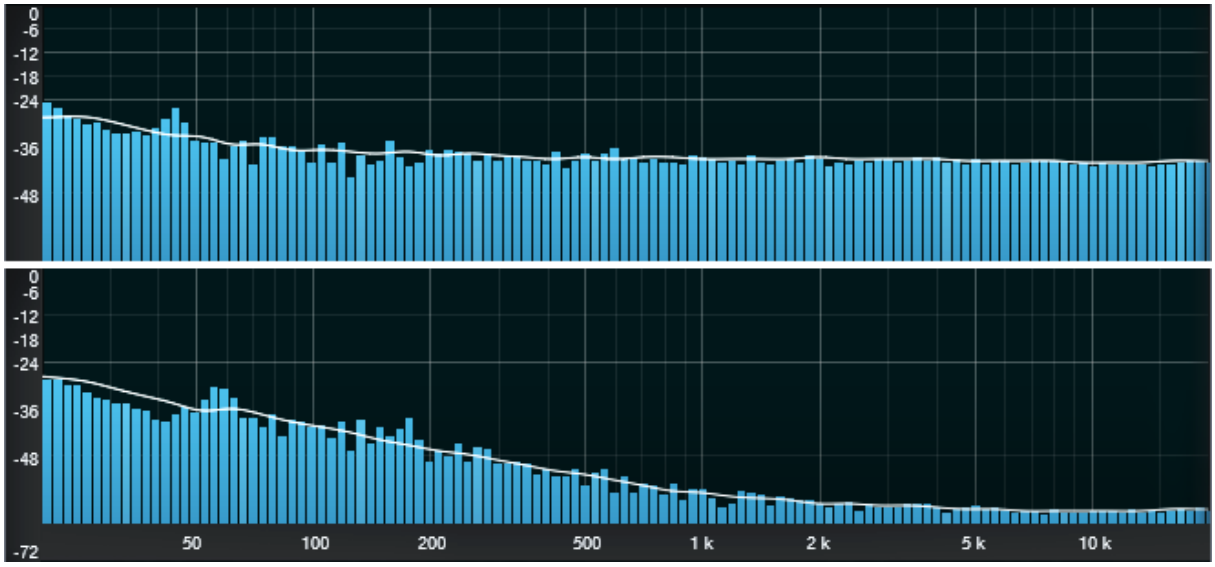
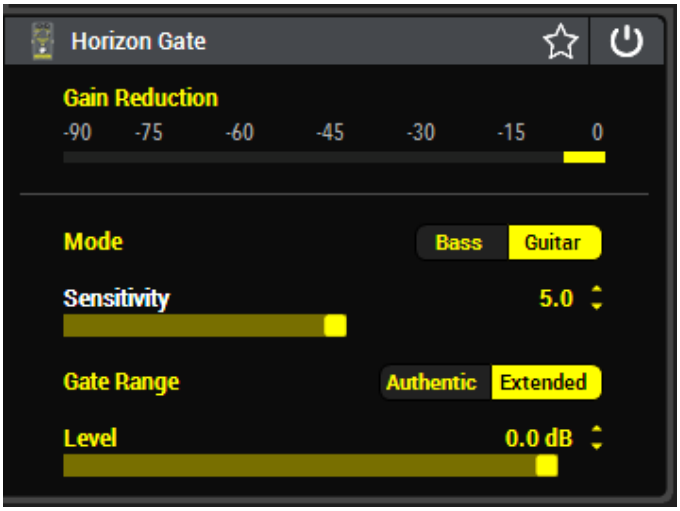


Figure 2.14 With an input level of -20 dB, the upper image shows a mild amount of high-frequency attenuation with Sensitivity set to 5.0. In the lower image, increasing Sensitivity to 10.0 produces much more high-frequency attenuation.

Although the controls are simple, they can be a little touchy to adjust. The settings in the screenshot provide moderate hiss reduction.

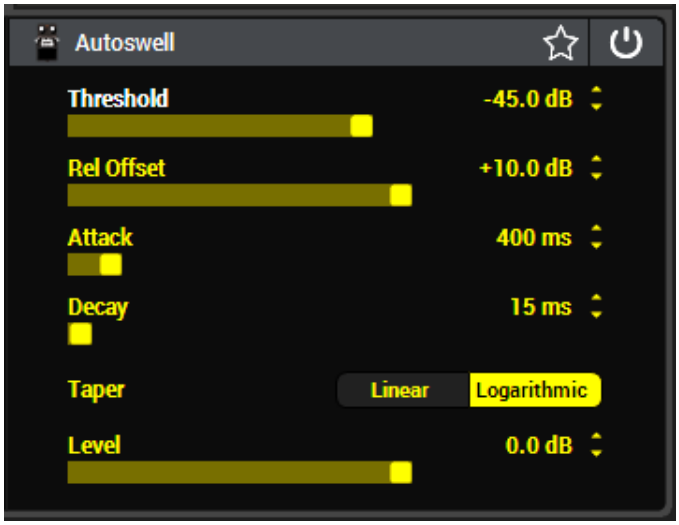
- **Mode** is self-explanatory—chose a response that's optimized for bass, or guitar.
- For a given input level, **Sensitivity** reduces the high frequencies more with higher settings.
- **Gate Range** sets a lower limit for how much the filter will attenuate the noise. With **Authentic**, the filter pushes the noise down around -30 dB. **Extended** increases this to -90 dB. (Note that the **Gain Reduction** meter is in Helix Native only, not in HX Edit or other Helix versions.)



The Horizon Gate typically inserts toward the beginning of a signal chain. However, it can follow a hissy amp so that when you're not playing, the hiss goes away. It can also follow a traditional noise gate so that when the gate's release time begins, the Horizon Gate reduces any residual hiss.

Autoswell

This effect is also known as attack delay and emulates bowed sounds. However, there's no "magic" setting that makes Autoswell work perfectly—you need to modify your playing technique to obtain the best results.



Plucking a note or playing a chord triggers Autoswell, which ramps up from minimum to maximum volume. To do its job, there needs to be a brief silence, or very low level, before the swell can begin. Autoswell works best when you mute the strings before plucking a note. To set up Autoswell:

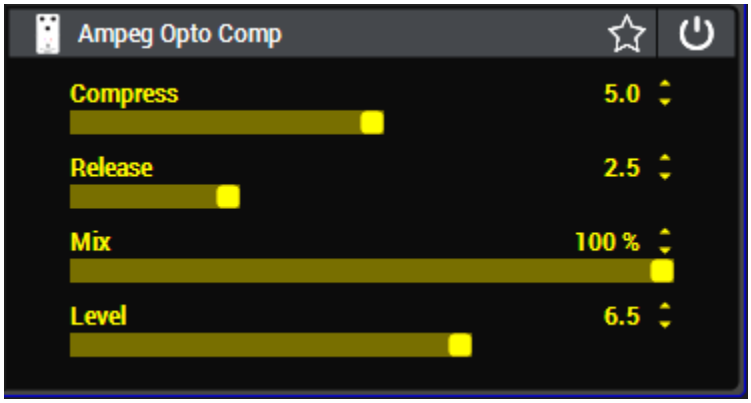
- **Threshold** is the most important parameter. When the guitar's level drops below the Threshold, the Autoswell action resets, and it's ready to ramp up again. You want the highest Threshold possible that can be exceeded by the notes you want to play. If you play a note that's not above the Threshold, the guitar will be silent.
- **Rel(ease) Offset** is like the Gate Close parameter on some noise gates. With more negative values, the note must decay to a lower level before the Autoswell action resets.
- **Attack** is the time it takes to ramp up from silence to full volume.
- **Decay** sets the time the volume will ramp back down to silence after the audio passes below the Rel Offset threshold. Longer settings allow notes to sustain longer, but then the Autoswell won't reset as rapidly. Use the minimum Decay necessary.
- **Taper** chooses a curve for the swell. A logarithmic curve more closely how the ear responds to dynamics, so the swell sounds smoother. The Linear curve sounds like it fades in and out faster at the fade's beginning. There's more information about logarithmic and linear hearing responses in the Volume/Pan block section.

Tip: If you're willing to play carefully, and mute strings between notes, Autoswell can be effective. I find using a volume pedal gives more predictable results, but a pedal can't go back to minimum as fast as Autoswell.

Ampeg Opto Comp

Compressors based on optical technology are smoother, and slower to respond to transients than their VCA- or FET-based counterparts (for more about optical compressors, see What Makes the LA Studio Comp Special). Optical compression technology is well-suited to bass, because the low frequencies may cause faster compressors to trigger on individual cycles. This often produces a "buzzy" sound. Increasing a compressor's attack control can minimize this buzz, but stompboxes often include only a decay or release control.

However, the Opto Comp isn't just for bass. With guitar, particularly for jazz and other types of music with a less intense guitar tone, Opto Comp gives a smooth, warm type of compressed character.



- **Compress** controls the compression ratio. 0.0 is a ratio of 1:1 (no compression). 5.0 is a compression ratio of 3:1, which many consider to be a "sweet spot" for bass (fig. 2.15). A setting of 10.0 gives 10:1 compression. Note that the threshold is fixed. If the Opto Comp doesn't appear to give enough compression, increase the level feeding the Opto Comp.

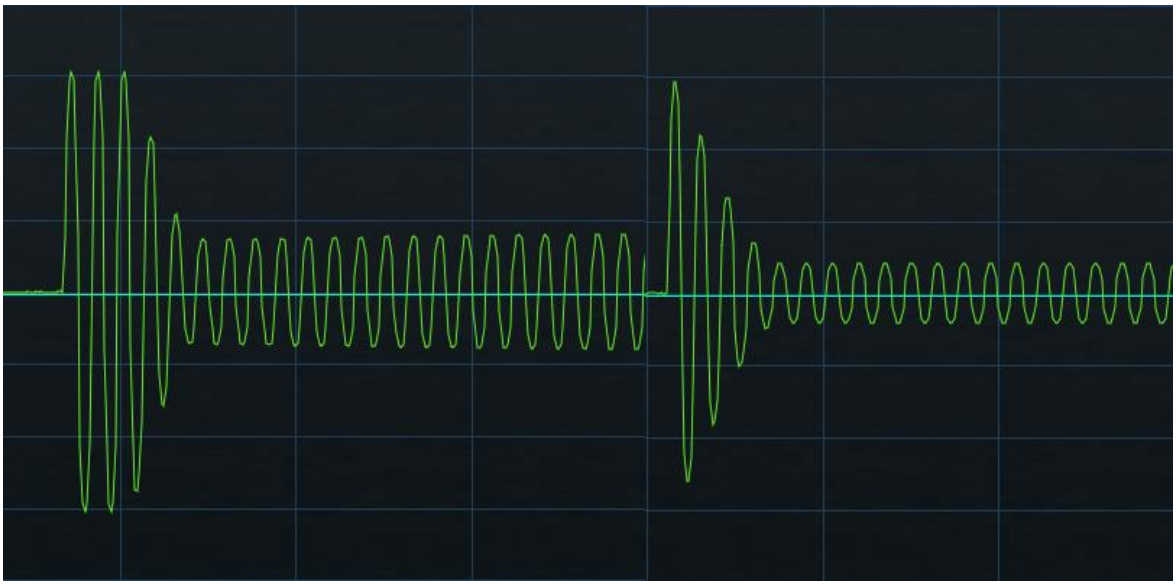


Figure 2.15 The attack on the left is with Compression set to 5.0 on a 100 Hz sine wave, while the one on the right uses a setting of 10.0. Note how the 10.0 setting clamps down faster on the attack and reduces the sustain level. The relatively slow attack gives a "smooth" punch. It also prevents any buzzing on low-frequency bass notes.

- **Release** sets the time it takes for gain reduction to go to zero after the input passes below the threshold. 0.0 gives a release of 75 ms, while the release at 10.0 is around 600 ms. This long release is part of what gives optical compressors a smooth sound.

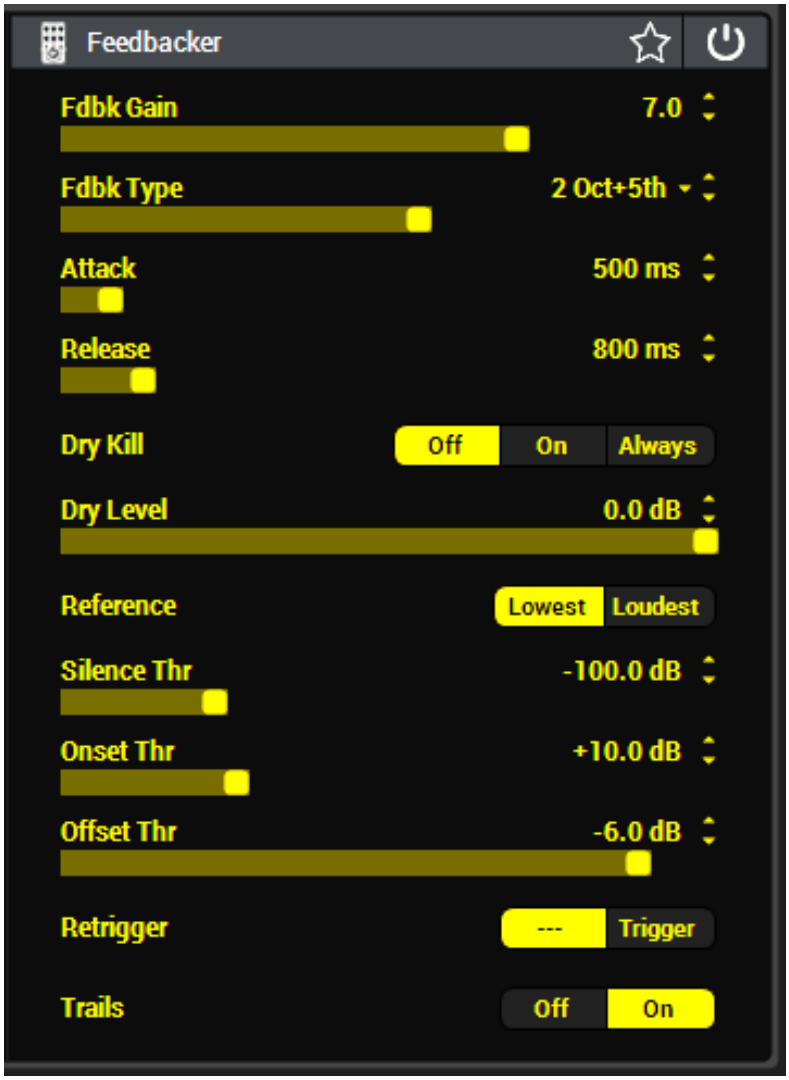
- **Mix** allows for parallel compression, because it controls the dry/wet mix. 0% is dry signal only, 100% is wet signal only.
- **Level** sets the block's output level, which is useful for gain-staging. It also provides makeup gain when using high compression ratios that reduce the output level.

Tip: Compression can reduce some of a guitar's sparkle, because lower frequencies typically have more level than higher frequencies. So, the compression affects high frequencies more than you might want. This is why some guitar sustainers include an internal filter to compensate for the loss of sparkle. Adding in some dry sound with the Mix control (e.g., 80 – 90%), or reducing Compress slightly, can restore some of that sparkle.

Feedbacker

This mono-only block works best as the first effect in a chain. Rare exceptions are preceding it with mild EQ or distortion. Like real feedback, the feedback effect can get out of hand if you're not in control of the settings. So, the Feedbacker loads bypassed by default.

When playing a chord, you can prioritize whether Feedbacker bases its feedback on the lowest-frequency or loudest note you play. This note becomes the Feedbacker's *reference frequency.* Any feedback intervals relate to this reference frequency. Feedbacker can also choose randomly among appropriate harmonics.



- **Fdbk Gain** controls the feedback amount. Set gain so that the feedback supports the guitar rather than competes with it. **Be careful!** Too much gain can make the feedback difficult to control. **Tip:** Assign the Feedbacker to a *momentary* footswitch, so feedback appears only while holding the switch.
- **Fdbk Type** chooses the feedback interval compared to the reference frequency. **Tip:** To add live performance versatility, assign different Feedback Types to snapshots.
    - **–Octave:** One octave below
    - **Unison:** Same as the reference frequency
    - **+Octave:** One octave above
    - **Oct +5th:** One octave plus a 5th above

- **+2 Octaves:** Two octaves above
- **2 Oct+3rd:** Two octaves plus a 3rd above
- **2 Oct+5th:** Two octaves plus a 5th above
- **2 Oct+7th:** Two octaves plus a 7th above
- **Mid to Low:** Feedback typically starts on the highest harmonic below 500 Hz. As the signal decays, the feedback drops down to lower harmonics.
- **High to Low:** Feedback typically begins on the highest harmonic below 1200 Hz. As the signal decays, the feedback descends to lower harmonics.
- **Rndm Onset:** A new attack (note or chord) triggers a randomly selected harmonic. Because the process is random, repeating the same chord may generate different harmonics.
- **Rndm Trigger:** Selects new harmonics randomly, based on the Retrigger parameter setting. For more details, see the **Retrigger** parameter info below.

- **Attack** ramps up the feedback level from 150 ms to 6 seconds.
- **Release** specifies how quickly each harmonic fades out, or transitions to a different harmonic. At higher release values, you may hear more than one harmonic during a transition.
- **Dry Kill (mute)** chooses how Feedback affects the dry signal:

    - **Off.** The dry signal is present when turning on the Feedbacker block. The Dry Level parameter (see below) controls the dry signal's level.
    - **On.** Turning on the Feedbacker Block mutes the dry signal.
    - **Always.** The dry signal never appears in the signal path, whether the Feedbacker is enabled or not. This is useful when the Feedbacker is in a parallel path.

- **Dry Level** sets the dry signal level through the Feedbacker block when **Dry Kill** is off.
- **Reference** prioritizes the reference note extracted from a chord. **Loudest** is the chord's loudest note. **Lowest** is a chord's lowest-pitched note (typically the fundamental).
- **Silence Thr** sets a threshold level. Signals below this level won't trigger feedback. Signals above this level can trigger feedback.
- **Onset Thr** matters only when **Feedback Type** is set to **Rndm Offset**. This parameter sets the threshold of onsets (plucks) that can change the feedback note:

    - **Lower values** increase sensitivity to plucking. The feedback note changes more often.
    - **Higher values** reduce sensitivity to plucking. The feedback note changes less often.

- **Offset Thr** controls the level below which the Feedbacker mutes the sustaining feedback note. Higher values inhibit warbling effects that can occur with overly low-level signals.
- **Retrigger** is intended to be assigned to an unused momentary footswitch. (Press and hold the Retrigger knob, press Learn, press the desired footswitch, and then set Type to "Momentary.") The generated feedback changes every time you press the switch. To confirm this, the parameter changes from "---" to "Trigger." How the feedback changes depends on the Feedback Type:

    - **Mid to Low** or **High to Low** types: Pressing the Retrigger switch causes the Feedbacker to descend to lower harmonics.
    - **Rndm Trigger** or **Rndm Onset** types: Pressing the Retrigger switch causes the Feedbacker to randomly choose a different harmonic.
    - **All other types:** Pressing the Retrigger switch regenerates feedback at the reference frequency.

- **Trails** turned off: Bypassing the block stops the feedback. **Trails** enabled: Feedback will ring out after bypassing the block, for the time set by the Release parameter.

Using Feedbacker

Feedbacker may seem difficult to control at first. However, you *can* make it do your bidding. Feedbacker requires practice to:

- Learn how the controls affect the sound.
- Find the optimum control settings for your playing style, the incoming level, and the desired result.

For example, how you set **Silence Thr** and **Offset Thr** can make the difference between smooth, controlled feedback that responds to your playing and a warbly, confused mess. **Tip:** Keep these parameters as high as possible, consistent with reliable triggering.

Feedbacker levels can change rapidly. Following it with the Rochester Comp (a leveling amplifier designed to maintain a constant level) can give more consistent levels. Check out the typical Rochester Comp settings in some of the following presets.

Processing Power

I often insert the Feedbacker block in a parallel path, with **Dry Kill** set to **Always** (no dry signal). Sometimes there's an amp in each parallel path. However, using two Amp blocks *and* the Feedbacker requires significant processing power. Fortunately, the Feedbacker sound by itself works well with most Amps. The Soup Pro is a good choice because of its low processing power requirements. That frees up power for other effects.

Obtaining Synthesizer-Like Effects

The Feedbacker was designed to complement an existing part, not stand on its own. Nonetheless, using the Feedbacker by itself (i.e., **Dry Kill** set to **Always**) can provide unique, synthesizer-like effects. This *requires* clean and precise playing:

- Mute strings you aren't playing.
- Avoid playing more than one note at a time.

When blending the Feedbacker with other sounds, either from being in a parallel path or by turning up the Feedbacker's Dry Level control, your playing technique doesn't need to be as clean.

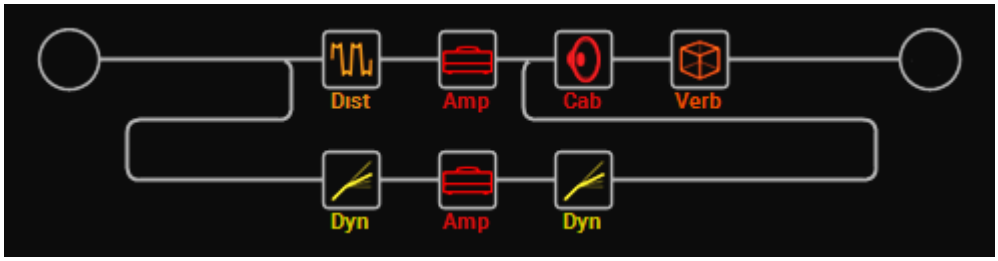
About the Presets

The *Free Files folder > Chapter 2 Presets > 3.70 Feedbacker Presets* includes new, custom presets that highlight the Feedbacker. For presets other than **FB Music Machine** and **FB Soundscape** (which are self-playing):

- ***Manage your expectations.*** Play slowly and cleanly for presets that expose the solo Feedbacker sound (**FeedbackrAmbient**, **Feedbacker Osc**, and **FeedbckrTheremin**). I wasn't sure whether to include these presets because they're difficult to tame. However, their novel sounds can work well in the studio. The presets that mix dry and feedback sounds are less demanding. In any case, when learning Feedbacker, be patient. Play how Feedbacker wants you to play.
- The **Silence Thr** and **Offset Thr** settings in the Feedbacker-based presets work well with my setup. If your playing doesn't trigger Feedbacker reliably, change **Silence Thr** to a more negative value. If the feedback cuts off too soon, or if new feedback notes mute preceding ones too quickly, change **Offset Thr** to a more negative value. *Experimentation is key* to accommodate your playing style and guitar.

- **Release** affects tracking. Shorter values may give better tracking.
- Tracking may be cleaner with humbucking and neck pickups.

Feedbackr60sRock

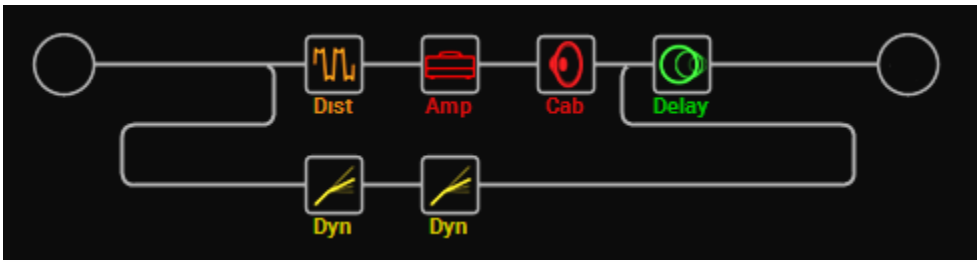


The feedback effect for leads has more clarity when using separate amps for the main guitar and feedback sounds. However, spending processing power on *two* amps limits which amps you can use. Remove the Legacy Verb to expand your amp choices, and/or have the option for a Dual Cab instead of a Single Cab.

The Rochester Comp tries to maintain a constant level for the feedback's audio. Its settings are crucial. Set the Threshold high enough so that the feedback attack time can develop before the leveling action kicks in. The Threshold value affects the output, so you'll need to adjust the Level whenever you change the Threshold.

Removing the Rochester Comp frees up a little more power, but the advantage of a consistent feedback level likely justifies leaving it in.

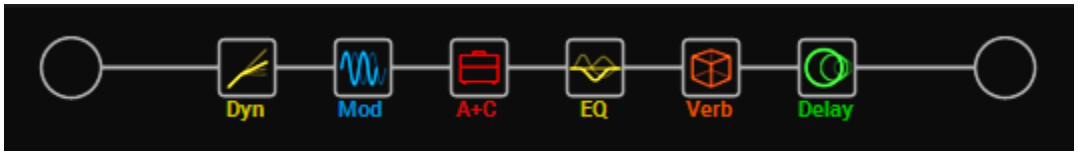
Feedbacker Lead



The Brit 2203 amp feeds a Dual Cab for a stereo image (which collapses well to mono). As with the previous preset, the Feedbacker is in a parallel path, and followed by the Rochester Comp. The same comments about the Rochester Comp parameters apply here.

The Delay provides ambience. Removing it frees enough processing power for most of the Legacy reverbs, or over half of the stereo (or most of the mono) Modulation processors.

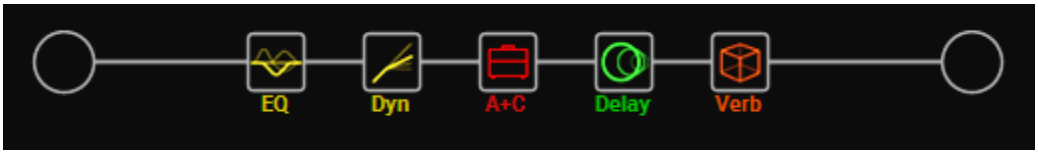
FeedbackrAmbient



This ambience-oriented preset is reminiscent of Greg Heet's E-Bow, and the "Frippertronics" effect popularized by Robert Fripp in the 70s. The Mod/Chorus Echo delay block has 4 seconds of delay with 92% feedback and chorusing, so there's a looping vibe too. The Feedbacker uses the Unison feedback mode.

Because the Feedbacker sound is solo (no parallel path or dry level), your playing must be super-clean. Also, play *slowly* to let the attack develop. Try playing notes on a single string, and sliding to different notes on that string. Often, you won't need to pluck the destination note, because sliding produces enough string vibration to keep the feedback sustaining. It's easy to get lost in this ambience-oriented, meditative preset.

Feedbacker Osc



This preset provides synthesizer-like waveforms (fig. 2.16). Different amps produce different waveforms. Different Drive settings change the waveform shape further.

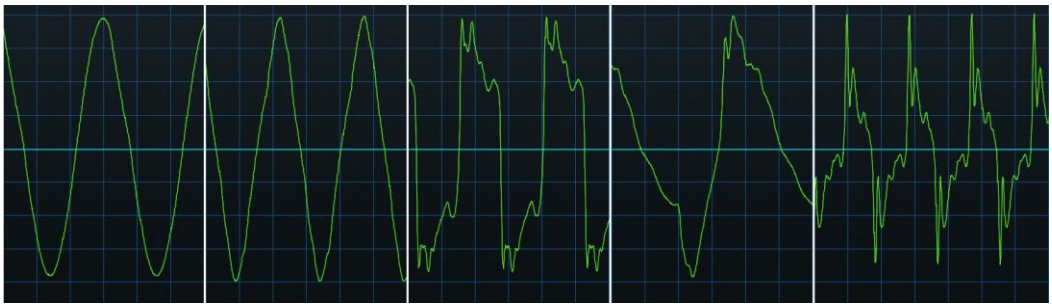


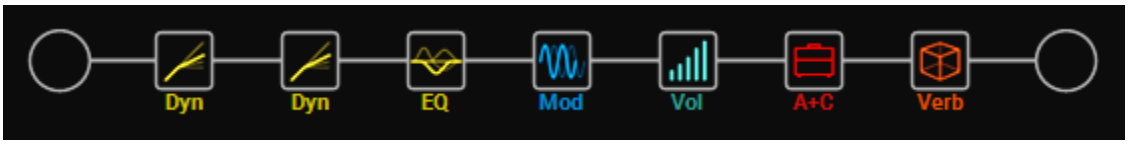
Figure 2.16 Left to right: Jazz Rivet (sine), US Double Vib (triangle), Kinetic (square), Mandarin 80 (sawtooth), Stone Age 185 (Pulse). Other amps produce other waveforms.

The Low and High Cut EQ at the beginning focuses on the guitar's frequency range. Choose the Amp+Cab based on the waveform you want. The Mod/Chorus Echo (Delay block) and Dynamic Hall (Reverb block) give a dreamy, ambient vibe. Replacing the Dynamic Hall reverb with a Legacy Hall reverb frees enough processing power to add most mono effects and some stereo ones.

Like other presets that expose the solo Feedbacker sound, you need to play cleanly. Try playing solos on a single string (high E, B, G) while muting the other strings. Sliding between frets on the string can produce portamento-type gliding effects.

Remember that higher **Silence Thr** and **Offset Thr** settings give fewer glitches. However, the optimum value depends on how hard and how consistently you play.

FeedbckrTheremin



The Theremin, invented over a century ago, is an electronic instrument based on a single oscillator. Waving your hand around the Theremin's antenna changed the oscillator's frequency. An additional plate that controlled volume also responded to hand motions. The Theremin produced some of the spooky sounds in 50s sci-fi and horror flicks. Assigning the Gain and Modulation Depth to EXP 1 provides more expressiveness. **Note:** You

won't hear any sound without pushing down on EXP 1. If your setup doesn't have an expression pedal, remove the EXP 1 assignments. Set the Vol block Gain to 0.0 and the Mod/Chorus Echo Depth to 50%.
This preset *requires* a specialized playing technique based on sliding between notes on a single string, with the other strings muted. The B and G strings seem to work best. The string needs to generate enough energy to drive the Feedbacker, hence the low **Silence Thr**. For the best results, push hard against the string as you slide, so that it vibrates more when it passes over a fret.

FB MusicMachine and **FB SoundScape**

You don't *need* a guitar for these, because both presets generate sounds on their own. The Feedbacker is set up for Random triggering. The 4 OSC Generator creates steady tones. Whenever a tone goes above the Feedbacker's **Silence Thr** parameter, the tone triggers feedback. The **Amp+Cab** varies the waveform and increases the level. The Legacy **Particle Verb** at the end enhances the effect with its own magic. **Note:** The Vol block in each preset links to EXP 1 so you can bring the sound in and out with the pedal. You will not hear any sound with EXP 1 all the way back in the heel position.

The presets have virtually identical parameter settings. **FB MusicMachine** is more rhythmic. **FB SoundScape** is more ethereal. Both are self-playing. However, OSC 4 mixes in Dry Level, so you can play guitar through them as well. Or, bypass the OSC 4 Generator to process only guitar.

The oscillator frequencies chosen for the presets are for the key of A. Unless you're processing only guitar with OSC 4 bypassed, playing in the key of A gives the most harmonious results.

Changing the level reaching the Feedbacker via one of several options varies the effect:

- **Bypass both Tremolos.** Assign the Gain block's Gain to an Expression Pedal. Vary the pedal to bring the effect in and out. The preset assigns the Gain parameter to EXP 1.
- **Enable the first Tremolo. Bypass the second Tremolo and Gain block.** This triggers tones automatically and rhythmically. Or, enable the Gain block, and then control when triggers occur with the Expression Pedal.
- **Bypass both Tremolos.** Assign the Gain block's Gain to an Expression Pedal. Vary the pedal to bring the effect in and out. The preset assigns the Gain parameter to EXP 1.
- **Bypass the 4 OSC Generator.** Trigger the feedback by playing your guitar.
- **Trigger from the 4 OSC Generator and guitar simultaneously.** The 4 OSC Generator's Dry Level is set to maximum, so any guitar playing at the input will pass through it.

This is just the tip of the iceberg:

- Change the Particle Reverb's **Condition** parameter to Critical.
- Change the Particle Reverb's **Condition** parameter to Hazard, Mix to 100%, and increase **Dwell**.
- For a pure, sine wave sound, choose the Jazz Rivet Amp+Cab. The level will be low. Set the Amp's Ch Vol to 10.0, Drive around 6.0 or 7.0, and the Cab's Level to +6.0 dB. Turn the EQ's output to +6.0 dB.
- For a harmonically complex sound, choose the **Oblivion** or **Fatality** Amp+Cab. Different amps and cabs give different waveforms.

EQ Blocks

The EQs in Helix are relatively conventional. Here are basic descriptions:

- **Simple EQ** has a high-frequency and low-frequency shelf, each with a **Gain** parameter, and a midrange boost or cut, with a fixed bandwidth (fig. 2.17). The Simple EQ requires very little processing power.

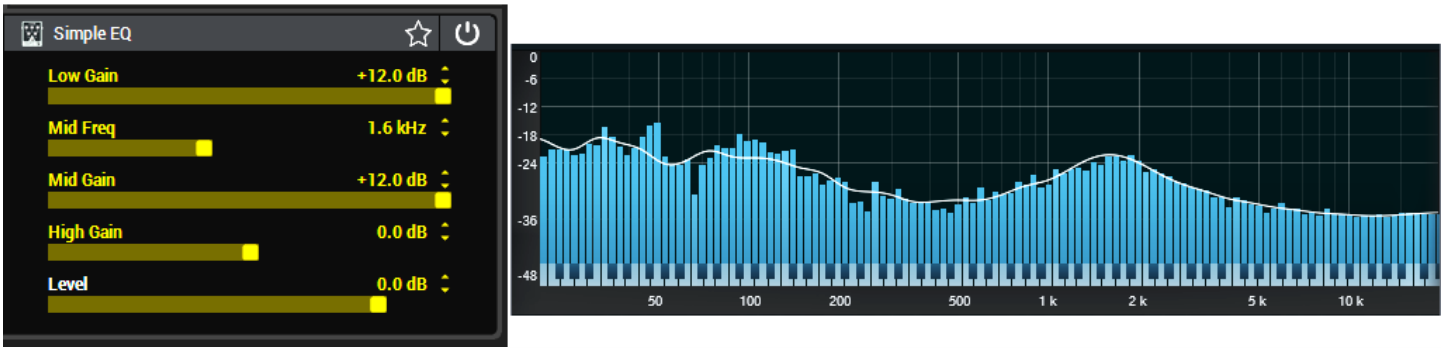


Figure 2.17 Simple EQ with settings and frequency response curve for a Low Gain boost, and Mid Gain peak.

- **Low and High Cut** (fig. 2.18) has 18 dB/octave filters that use **Cut** parameters to attenuate high and/or low frequencies. The high and low frequency limits respectively for the Low and High filters is 1 kHz.

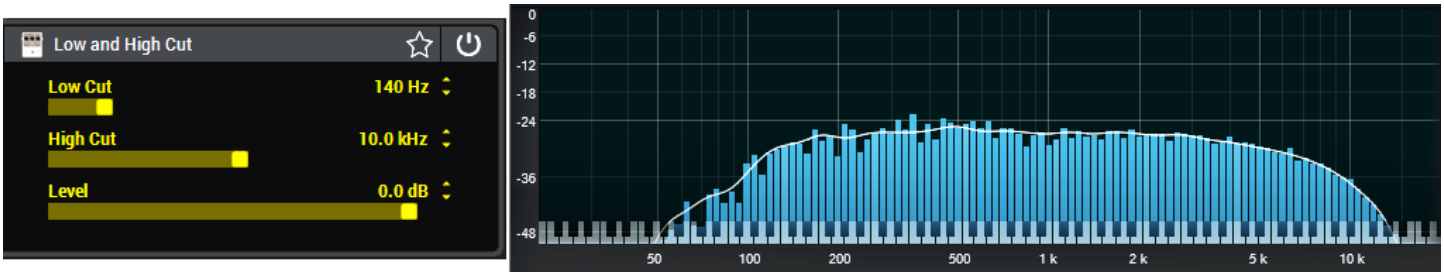


Figure 2.18 The frequency response curve confirms that the Low Cut filter is rolling off lows at 140 Hz, while the High Cut is rolling off highs at 10 kHz.

- **Low/High Shelf** (fig. 2.19) uses conventional shelf filters. These have **Gain** parameters to boost or cut starting at a specific frequency, but the response levels off eventually. The **Low Freq** covers 20 Hz to 2.0 kHz. The **High Freq** covers 400 Hz to 15 kHz.

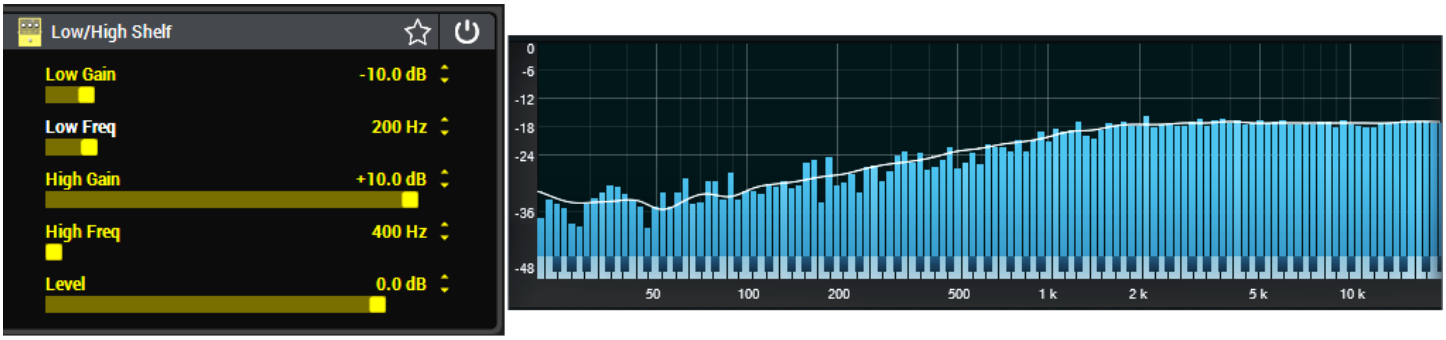


Figure 2.19 The High Shelf is boosting, starting at 400 Hz. Note in the frequency response graph how the boost levels off around 2 kHz. Meanwhile, the Low Shelf is cutting frequencies below 200 Hz.

- **Parametric** (fig. 2.20) is precise and flexible. It includes **Low Cut** and **High Cut** filters, as well as three parametric stages. Each stage has adjustable **Frequency**, **Gain** (boost/cut), and **Q** (resonance).

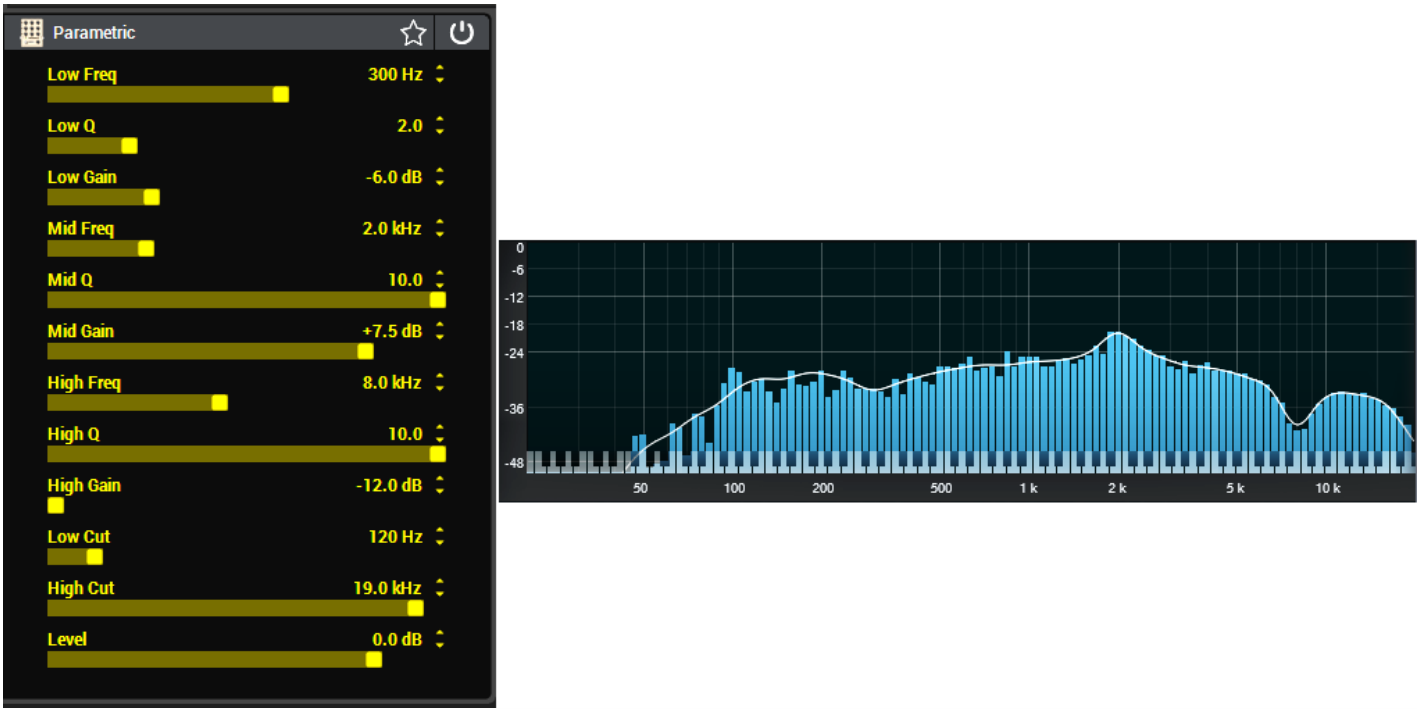


Figure 2.20 The frequency response curve and settings show a low cut starting at 120 Hz, a moderately shallow cut at 300 Hz, a strong, narrow midrange peak at 2 kHz, a deep high frequency notch at 8.0 kHz, and a high cut at 19 kHz.

- **Tilt** (fig. 2.21) starts amplifying or reducing the high or low frequencies at a particular **Center Freq**. For example, the Dark **Tilt** setting increases the lows, and attenuates the highs, as the audio moves further away from the Center Freq. The Bright **Tilt** setting increases the highs, and attenuates the lows, more as the audio moves further away from the Center Freq. Use Dark to give single-coil pickups a humbucker character, and Bright to give humbuckers more of a single-coil vibe.

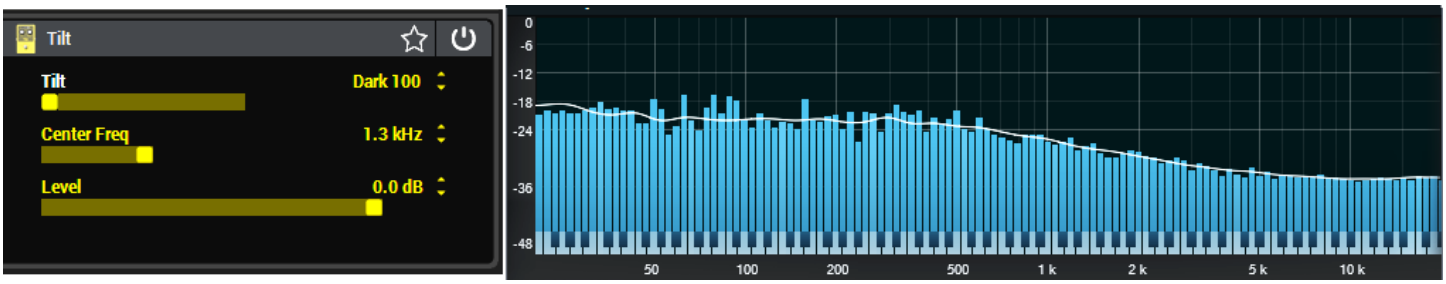


Figure 2.21 With a Dark tilt and a 1.3 kHz Center Freq, the low frequencies increase while the highs decrease.

- **Cali Q Graphic** (fig. 2.22) is a 5-stage graphic equalizer. A slider for each band (**80**, **240**, **750**, **2200**, and **6600** Hz) sets that band's level. There's also a master **Level** parameter. This EQ shapes tone in a relatively gentle way, so it can coax a speaker cab into sounding more like other cabs. However, the EQ on which this is modeled is probably best known for creating the scooped midrange sound that's common with high-gain amps, specifically those designed for metal. Note that this kind of tone-shaping can also be useful with keyboard pads, drums, and even stereo program material.

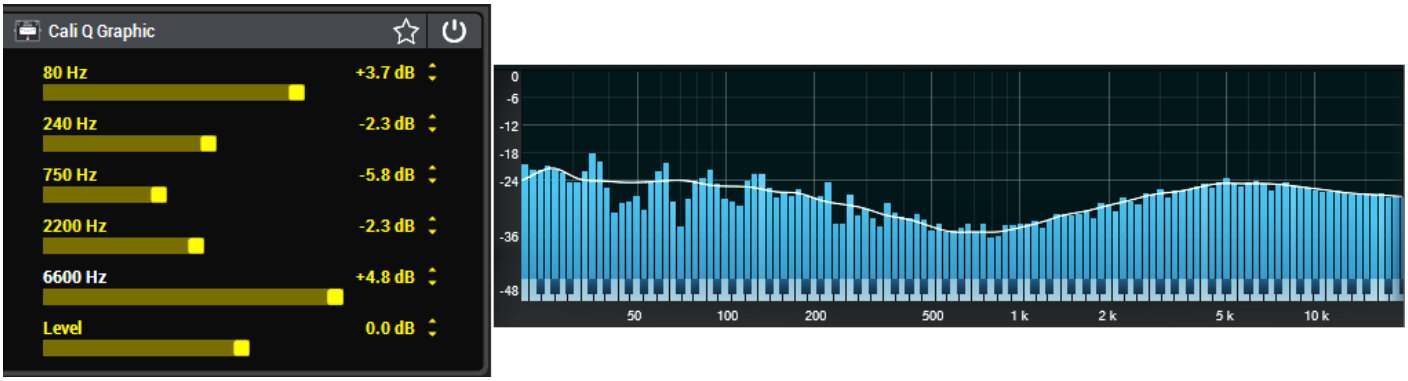


Figure 2.22 These Cali Q Graphic settings create a V-shaped frequency response scoop.

- **10 Band Graphic EQ** (fig. 2.23) is a more flexible version of typical stompbox graphic EQs. A slider for each band (**31.25 Hz**, **62.5 Hz**, **125 Hz**, **250 Hz**, **500 Hz**, **1 kHz**, **2 kHz**, **4 kHz**, **8 kHz**, and **16 kHz**) sets that band's level. There's also a master **Level** parameter. Like all graphic EQs, the tone-shaping is fairly gentle. With 10 bands, this EQ can create complex response curves over a wide frequency range.

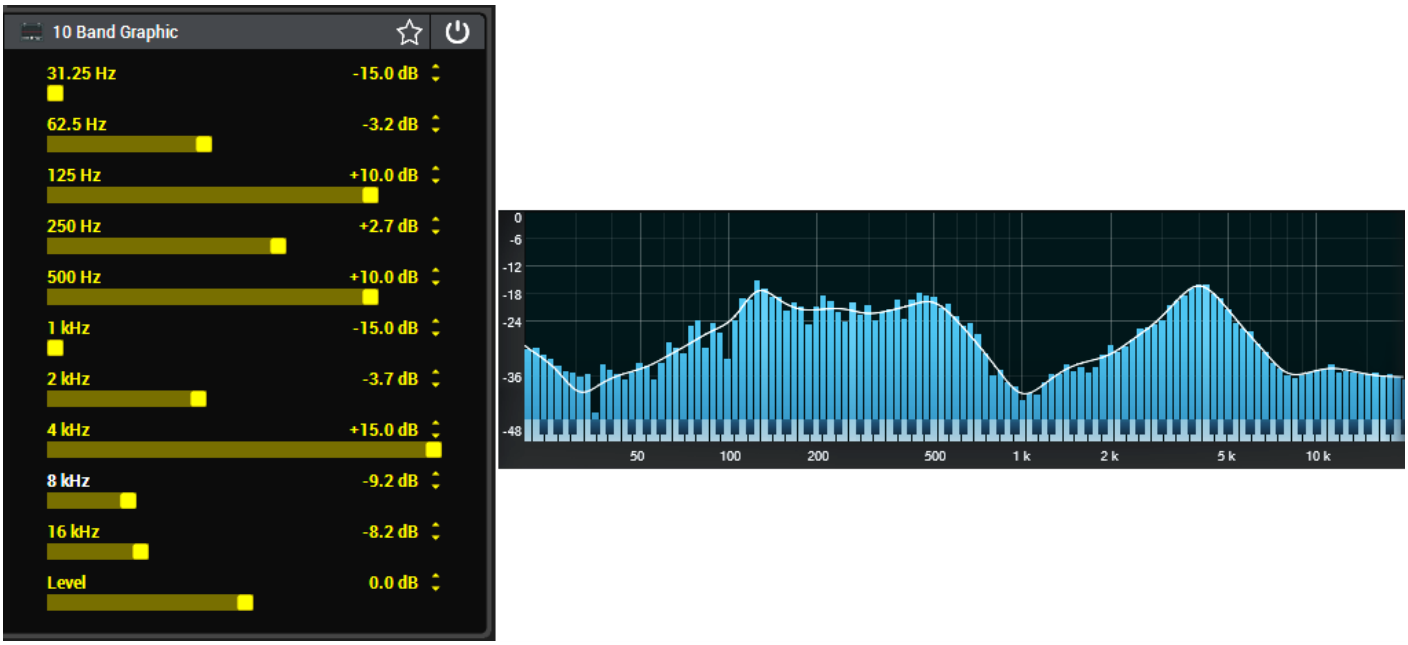


Figure 2.23 For general tone-shaping over the full audio range, the 10 Band Graphic EQ can create simple, or sophisticated, EQ curves. The frequency response curve reflects the settings in the screenshot.

Acoustic Sim

This is an EQ block, but it has a different purpose that merits its own section. The Acoustic Sim's goal is to impart a more acoustic-like sound to electric guitars. The *Free Files > Chapter 2 Presets* folder includes the preset **Acoustic Sim**. It's a decent place to start, but the ultimate results depend on your guitar, pickups, strings, and playing style.

Manage your expectations—an electric guitar won't sound like an acoustic guitar. They have different body types, use different strings, and place the strings under different amounts of tension. Miked acoustic guitars introduce additional variables from the mic, mic position, and room acoustics. However, an electric guitar can sound like an electrified acoustic guitar that's going direct. For gigs with a couple acoustic songs, and you don't want to bring another guitar, acoustic simulation can provide a substitute.

Also note that you can load acoustic guitar IRs into an IR block. Some will prefer using IRs over using the Acoustic Sim.

Settings

The Acoustic sim (fig. 2.24) has four main sets of controls.

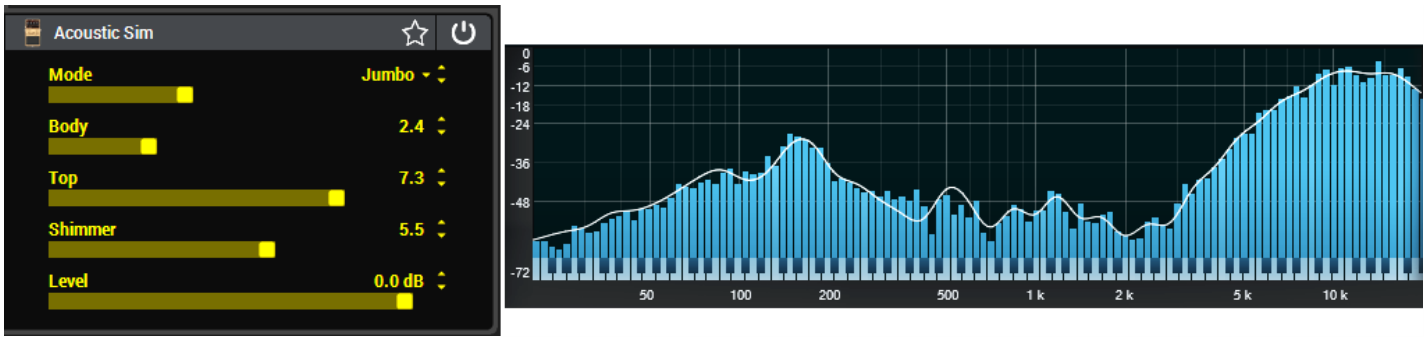


Figure 2.24 Acoustic Sim, with settings that worked well with my guitar's neck humbucker.

Four modes choose the guitar type:

- **Standard** (dreadnought body style)
- **Jumbo** acoustic guitar body
- **Enhance** (emphasizes attacks)
- **Piezo** to emulate Ovation-type round back guitars

The additional controls alter timbre:

- **Body** alters the level of resonances below about 2 kHz, so higher Body values give a bassier sound. The resonances for different guitar types change, but altering Body doesn't alter a given body's resonances.
- **Top** emphasizes highs. Acoustic guitars sound very bright compared to electric guitars, so the treble boost is massive. Be careful that the high frequencies don't overload subsequent stages.
- **Shimmer** produces rapid, subtle amplitude fluctuations starting at around 2 kHz. The higher the value, the greater the level fluctuations.

Much of the Acoustic Sim's effectiveness depends on your pickups and playing style. Neck and middle pickup positions, using single-coil pickups, work best for me.

Modulation Blocks

Many of Helix's modulation effects are familiar. Some effects, such as tremolo, have been around for over half a century. So, like the section on Distortion, this section places more emphasis on effects that are new, different, or have unusual parameters that might not be familiar to all users.

Let's begin with a refresher on modulation effect parameters. Modulation involves changing some parameter's value in response to a different parameter. For example, traditional tremolo changes an amp's level, according to the level and frequency of a low-frequency oscillator (LFO). Here are the most common parameters you'll find in modulation effects:

- **Modulation or LFO Width.** With flanging, vibrato, and chorusing, modulation varies an initial delay time cyclically. The delay time varies between a higher and lower limit (the *width* or *range*), which causes the pitch to vary between sharp and flat. A wide sweep range is necessary for dramatic flanging. Chorus and echo don't need much sweep range to be effective. With tremolo, modulation changes the level (amplitude) between a maximum and minimum level.
- **Modulation Waveform (Shape).** The modulation source usually comes from low-frequency waveforms (triangle, sine, square, and sawtooth waves). Other possibilities are random modulation waveforms, or modulation that steps sequentially through various levels.
- **Waveform Duty Cycle** alters the wave shape (fig. 2.25). A sine wave with a 50% duty cycle takes 50% of the cycle to go from zero to the maximum value (positive or negative), and 50% of the cycle to go from the maximum value back down to 0. With a 25% duty cycle, it takes 25% of the cycle to go from zero to the maximum value, and 75% of the cycle to go from the maximum value back down to 0. With a 75% duty cycle, it takes 75% of the cycle to go from zero to the maximum value, and 25% of the cycle to go from the maximum value back down to 0.

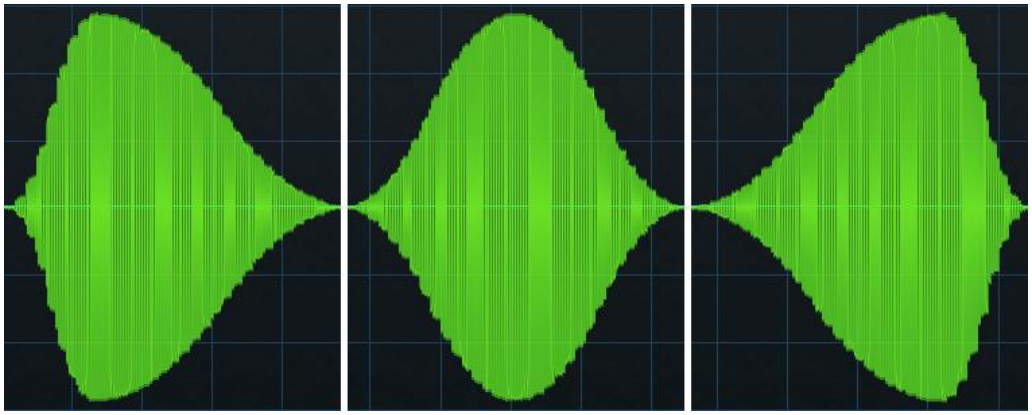


Figure 2.25 Tremolo modulation with a sine wave. Left to right: 25% duty cycle, 50% duty cycle, and 75% duty cycle.

- **Modulation Speed (Rate)** sets the modulation frequency. Helix can sync most modulation effect rate parameters to tempo generated by a MIDI clock, so variations happen in time with the rhythm.
- **Modulation Intensity (Depth)** injects more or less modulation into the processed signal. For example, vibrato sounds best with light modulation amounts. Increasing depth sounds more like a warped record than vibrato.
- **Spread.** With modulation blocks, Spread alters the modulation's phase relationship between the left and right channels. At 0.0, the modulation signals are in phase. At 10, the modulation signals are 180 degrees out of phase. At 5.0, they're 90 degrees out of phase. In simpler terms, higher values increase a stereo image's width, so it covers a wider range between the stereo image's left and right extremes.
- **Mix (Wet/Dry)** alters the balance between a processed and dry signal.

- **Level** sets the block's output level, which allows for gain-staging.
- **Voices.** Some chorus units split the guitar into more than just one dry and one processed path. Each processed path is called a *voice*. For lush chorusing effects, additional voices diffuse the sound more.
- **Initial Delay.** You'll find this with time-based modulation effects, like chorusing and flanging. Initial delay sets the base delay time, generally between 0 and 20 ms. With flanging and chorusing, modulation occurs around this initial time delay. Flanging typically uses shorter initial delay times (0 – 9 ms). Chorusing uses longer initial delays (5 – 20 ms).
- **Feedback** (also called **Regeneration**) feeds some of the output back to the input. This produces a sharper, more resonant sound with flanging and phasing (somewhat like increasing a filter's resonance control). The feedback's polarity may be positive or negative. Positive feedback gives a sharper sound with a more defined sound of pitch, while negative feedback has a more whooshing, hollow character.
- **Headroom** is a unique Helix parameter that comes into play with hotter-than-normal input signals (fig. 2.26). At 0, the output reflects any overload occurring with the effect, as caused by too high an input level. Negative headroom values bring peaks more under control, like adding saturation. The advantage of having a headroom parameter is being able to control any peaks inside the effect, instead of elsewhere in the effects chain. Positive headroom values open up more headroom (dynamic range), which reduces distortion by being more tolerant of overload conditions.

Figure 2.26 Left: Headroom at 0 (note a bit of saturation). Middle: Headroom at -12 shows heavier saturation. Right: headroom at +12 shows no saturation. These settings are from the Elephant Man delay.

Optical vs. 60s Bias Trem

These have the same basic controls: **Speed**, **Intensity**, **Level** and for the stereo version, **Spread**. The 60s Bias Trem also incorporates a vibrato (periodic frequency change) option.

The difference is the LFO waveform they generate. The Optical Trem has a more jagged sound, because the amplitude ramps up slowly, then drops back quickly (fig. 2.27). Decreasing Intensity gives a more subtle effect but doesn't alter the fundamental character.

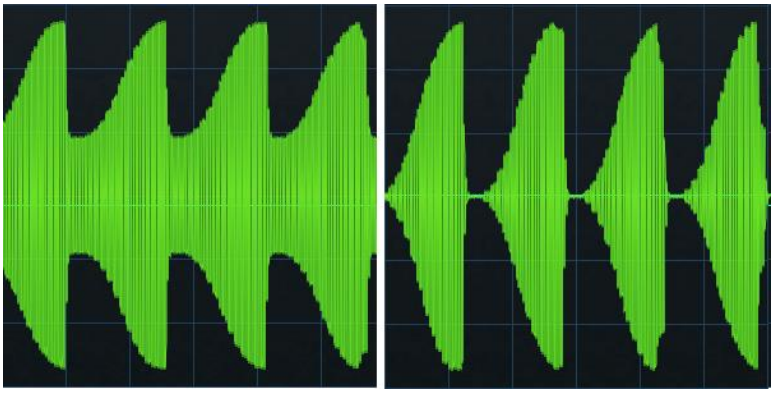


Figure 2.27 Optical waveform with medium Intensity (left) and maximum Intensity (right).

The 60s Bias Trem has an even, symmetrical waveform (fig. 2.28) whose amplitude ramps up and down equally. This gives a smoother tremolo effect, which is also important for vibrato.

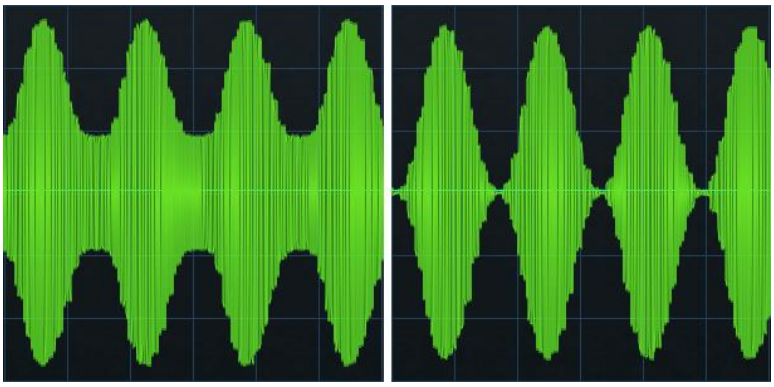
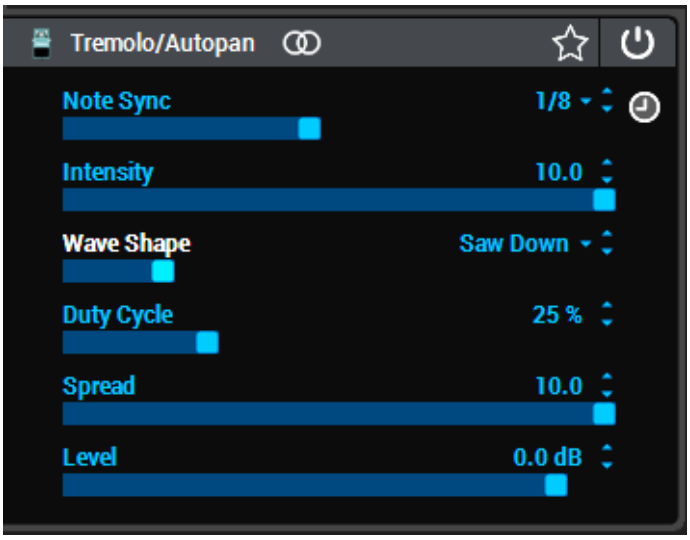


Figure 2.28 60s Bias Trem with medium Intensity (left) and maximum Intensity (right).

Tremolo/Autopan

This Tremolo/Autopan offers more control over the tremolo effect than the Opto or 60s Bias Trem effects.



There are four familiar controls: **Rate** (shown set to Note Sync for synching to tempo), **Intensity**, **Spread** for stereo, and **Level**. **Wave Shape** offers seven synthesizer-type LFO waveforms (fig. 2.29), not just the usual sine wave, to modulate amplitude. Note that Inverse Sine is most effective with Intensity settings of 9.9 or lower.

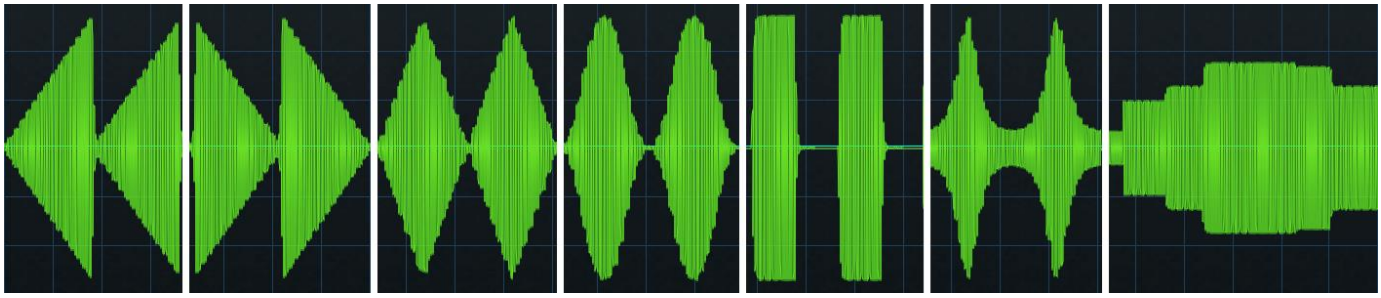


Figure 2.29 Left to right: Saw Up, Saw Down, Triangle, Sine, Square, Inverse Sine, and Random.

Duty Cycle further alters the wave shape, as described in Modulation Blocks.

Panning

The stereo version becomes an autopanner instead of a tremolo. The panning action depends on the input:

- With a mono input, the mono sound pans between the left and right channels.
- With a stereo input, the panner acts like a balance control. When the panner pans left, you'll hear the left channel input. When it pans right, you'll hear the right channel input. When the pan position passes through center, you'll hear the left input in the left channel, and the right input in the right channel.

Rhythmic Tremolo Possibilities

The many Wave Shapes can create powerful rhythmic effects. These are well-suited for EDM-style music. Here are some suggestions, which are especially useful with note sync:

- Insert two tremolos in series. Set the first tremolo to a fast Rate, and the second one to a slow Rate. The second tremolo will let through "pulses" of sound from the first tremolo. The **Dual Tremolo** preset in the *Free Files > Chapter 2 Presets* folder works this way.
- A slow **Saw Up** wave sounds somewhat like backwards tape.
- Synching **Saw Down** to notes gives percussive guitar stabs. This is particularly effective when synched to the beat. 1/8th notes are a good start.
- The **Square Wave Shape** gives "slicing" effects—like toggling a pickup switch between the bridge pickup, and the neck pickup with its level all the way down.
- Helix doesn't necessarily know where a beat begins. If the Note Sync effect seems out of sync, tap the tempo button briefly to re-sync.

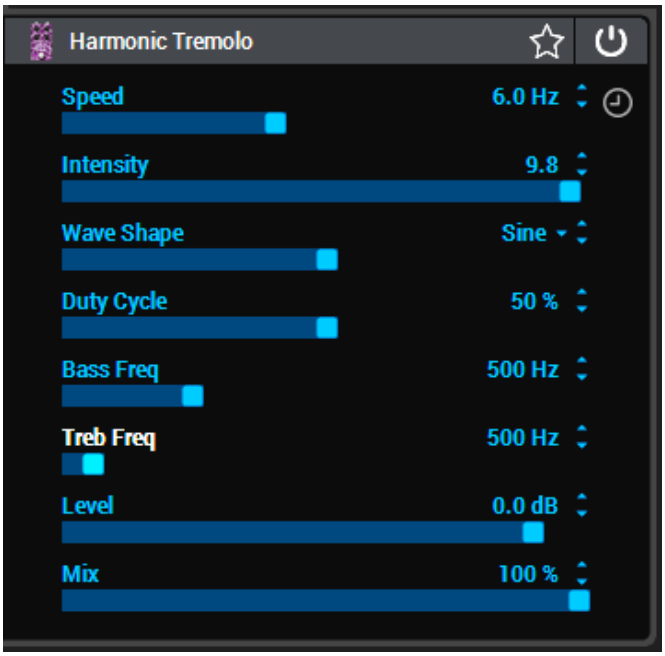
Harmonic Tremolo

The Helix Harmonic Tremolo is more flexible than the original Harmonic Tremolo circuit found in some early "brown" amps. To understand the improvements, it helps to understand how the original worked.

The input signal splits into high and low frequency bands, and then an LFO amplitude-modulates them out of phase so that while the highs get louder, the lows get softer and vice-versa. The preset **OrigHarmonicTrem** in the *Free Files > Chapter 2 Presets* folder is in the spirit of the original, pre-Helix circuit.

The Helix version is similar to the Tremolo/AutoPan. Please refer to it for explanations of all parameters except for Bass Freq and Treble Freq:

- **Bass Freq** sets the upper cutoff for the low band.
- **Treb Freq** sets the lower cutoff for the high band.

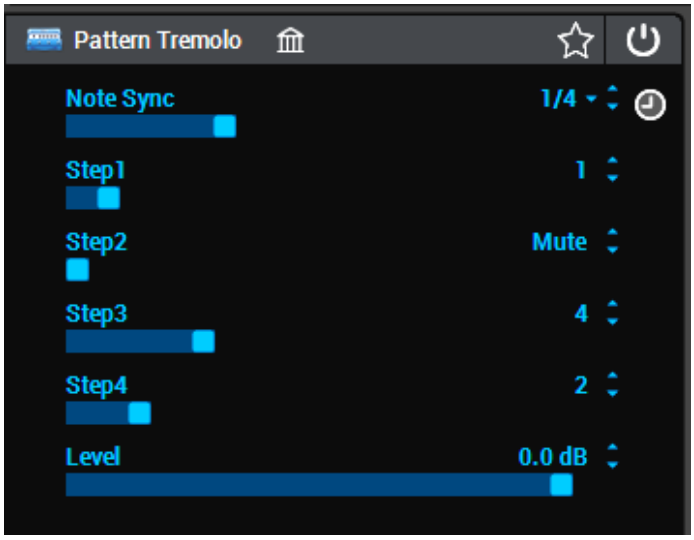


Altering these two parameters gives a wide range of harmonic tremolo sounds. Spreading them further apart creates a notch when the Harmonic Tremolo crosses over from the high band to the low band, or vice versa. For example, setting the Bass Freq to 250 Hz and the Treb Freq to 1 kHz produces a notch around 500 Hz.

Setting the Bass Freq higher than the Treb Freq creates a slight boost at the crossover point.

Pattern Tremolo

Technically speaking, all tremolos have a pattern—but this produces a 4-step sequence of rhythmic patterns. The Bleat Chop Trem is a more advanced implementation of this effect, but knowing how the Pattern Tremolo works makes it easier to understand the Bleat Chop Trem. Although the control set seems simple, the interactions between the **Steps** can be complex.



- Each step is equal to the note sync rhythm when synched to tempo, or the period when set to ms.
- Each step generates a square wave (on/off) modulation source. This varies the level between on and off.
- You can divide each step into a specified number of square wave "slices," from 1 to 16. You can also choose **Mute** (the step is off, which mutes the signal going through the Pattern Tremolo) or **Full**, where the step is on full, so the audio isn't chopped by the square wave.

For this, a picture is indeed worth a thousand words—see fig. 2.30. This shows the settings shown above in the screenshot and colorizes them for clarity.

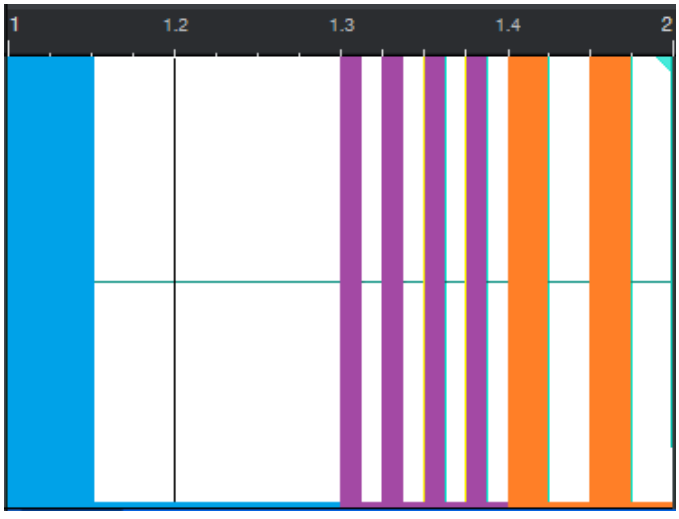


Figure 2.30 The Pattern Tremolo in action. Step 1's modulation is blue, step 3 is purple, and step 4 is orange.

According to the Note Sync parameter, each step is a quarter-note, and there are four steps. So, the pattern is one measure long.

- The first step (blue) has a value of 1, so there's one square wave that lasts for a quarter-note. This creates one slice. Because the modulation signal is a square wave that alternates between a maximum and minimum level, the audio is on for an eighth note, and off for an eighth note.
- The second step is Muted. So, the measure's second quarter-note remains off, and you don't hear any audio.
- The third step (purple) has a value of 4, so there are four slices within the third beat. Each slice is a 16th note, so the audio is on for a 32nd note, and off for a 32nd note.
- The fourth step (orange) has a value is 2, so there are two slices within the fourth beat. Each slice is an eighth note, so the audio is on for a 16th note, and off for a 16th note.

At slow tempos, you can hear the discrete on/off changes easily. Higher step values, like 16, produce a burst of pulses that's pretty cool.

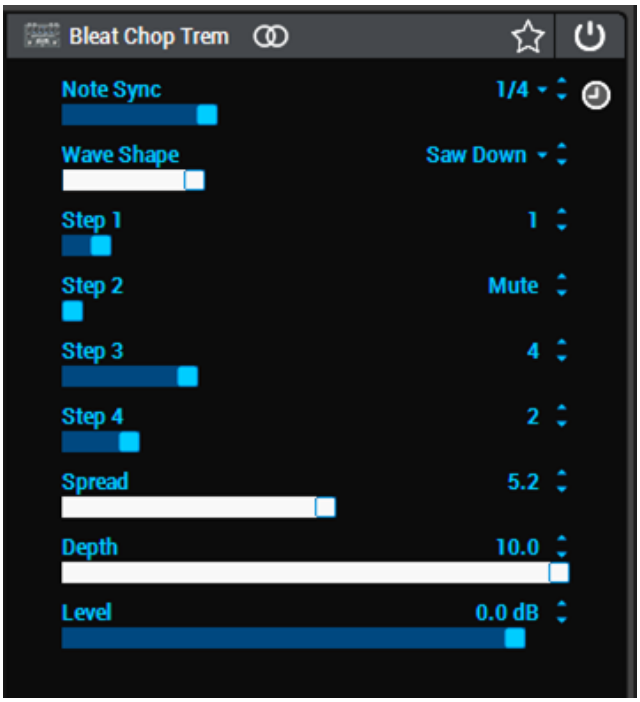
Limitations

The Pattern Tremolo is free-running. So, there's no guarantee that step 1 will start at the beginning of a measure when you enable the effect. This is true even when synched from an external source, like a DAW. With a hardware Helix, hit the Tap button briefly on the downbeat to reset the pattern to its beginning.

With Helix Native, this lack of sync is a problem, because there's no Tap Tempo function. It's hit or miss whether the pattern will sync correctly to your DAW. The workaround is to use Helix as an audio interface. Send a DAW's track through a hardware Helix with the desired preset loaded, use the Tap Tempo button to sync the sequence to the music, and then record the result in your DAW.

Bleat Chop Trem

The Bleat Chop Trem improves upon the Pattern Tremolo with three additional parameters (highlighted in white below), although the basic principle is the same. Please refer to the Pattern Tremolo for more information.



- **Wave Shape** provides not just a square wave, but Sawtooth Up, Sawtooth Down, Triangle, Sine, and Square. Triangle and sine avoid abrupt on-off transitions, while Sawtooth up creates an attack time, and Sawtooth Down, a decay time. Sync has the same limitation mentioned for the Pattern Tremolo.
- **Depth** also gives more modulation flexibility, because this parameter determines how much the Bleat Chop modulation affects the audio.
- **Spread** is only in the Stereo version. The same pattern repeats in the left and right channels (fig. 2.31), but Spread alters the LFO waveforms' phase relationship, as described previously.

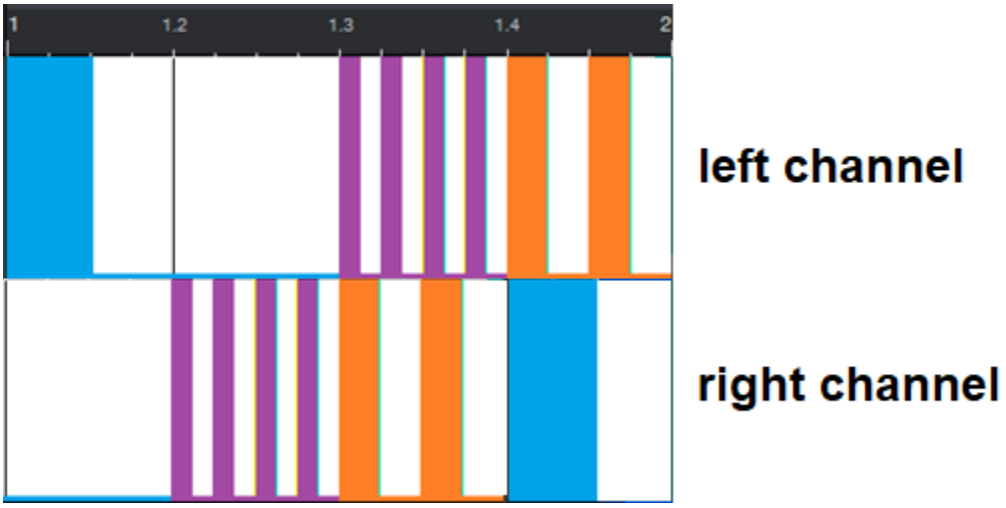


Figure 2.31 This uses the same settings shown in the Pattern Tremolo screenshot (fig. 2.30), along with a square Wave Shape, but uses the Bleat Chop's stereo mode.

Ubiquitous Vibe

This is sonically like a combination phase shifter and chorus. **Rate**, **Intensity**, **Mix**, and **Level** are all familiar controls. The screenshot shows typical parameter values for Chorus (left) and Vibrato (right), but **Lamp Bias** requires an explanation because its setting makes a huge difference in the sound.

The original unit used a lamp in conjunction with photo-resistors to change the phase. Lamp bias was a trimpot that set the current going to the lamp, which the LFO varied. Changing the bias altered the way the lamp cycled between dim and light. If the lamp went from dim to bright rapidly (a high Lamp Bias value in Helix), the sweep is more on/off than smooth. Reducing the Lamp Bias value creates a smoother-sounding sweep. Of course, you'll end up adjusting the setting with your ears, but fig. 2.32 gives a visual idea of what to expect. Note that low values decrease the output level, so compensate with the Level parameter.

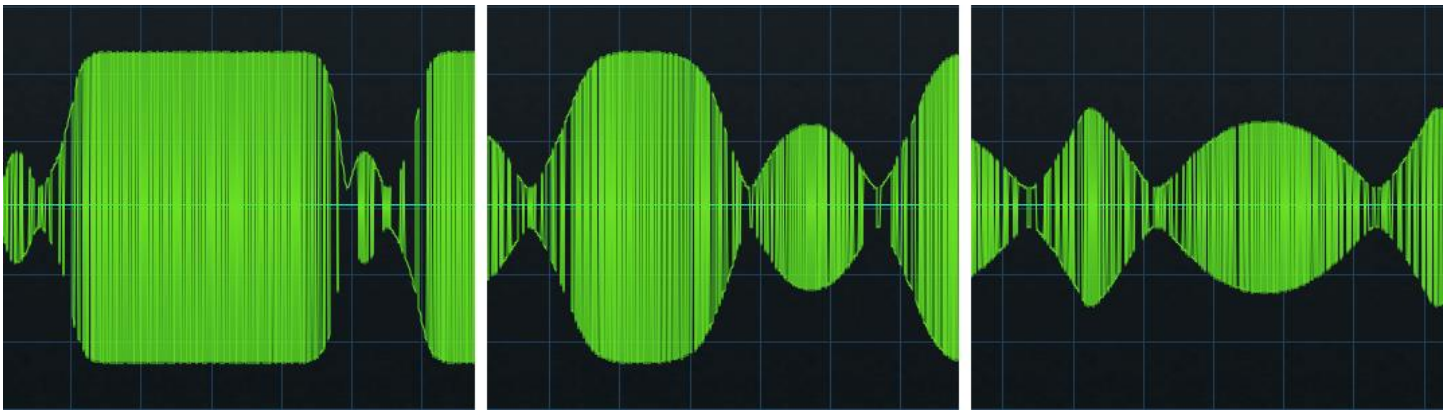


Figure 2.32 Left to right: Lamp Bias values of 8.0, 3.0, and 1.0, with maximum Intensity and slow rate (1.3).

The Ubiquitous Vibe vibrato is subtly different from Helix's other vibrato options. For this application, I prefer low Lamp Bias values—no higher than 2.0, and usually 1.0.

Script Mod Phase and Pebble Phaser

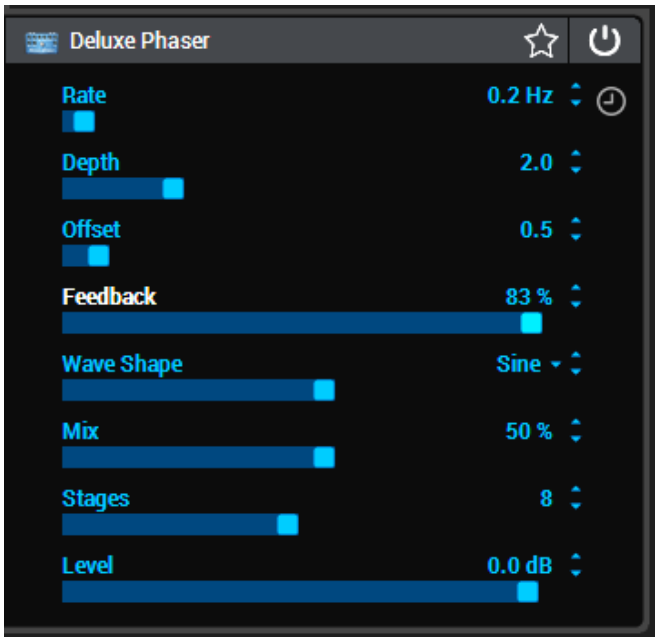
The Script Mod Phase and Pebble Phase are basic. With both blocks, **Rate** controls the speed, and **Level** sets the output level. With the Pebble Phaser, **Color** changes the tone.

The Script Mod Phase's **Mix** parameter alters the dry/wet mix, with the most intense flanging occurring around 50%. Higher settings tend more toward vibrato.

For a Script Mod Phaser application, see the Combi-Band Phaser preset. This parallel processing preset takes advantage of the Script Mod Phase's simplicity.

Deluxe Phaser

The Deluxe Phaser has a more comprehensive control set than the previous phasers.



Rate, **Mix**, and **Level** are familiar. Here are details on the other parameters:

- **Depth** sets the sweep's high-frequency limit. If you set this too high, the sound can become "thin" at the top of the sweep.
- **Offset** sets the sweep's low-frequency limit. If you set this too low, the sound can be "tubby" at the bottom of the sweep.
- **Feedback** increases the phasing effect's intensity. 0 (at the slider's center) produces minimal intensity. With positive feedback, the phaser's character is slightly thinner and sharper than negative feedback's somewhat fuller sound.
- With higher values, the **Stages** parameter increases the phasing effect's sonic complexity.
- **Wave Shape** (fig. 2.33) alters the sweep's shape, with the same options as the Tremolo/Autopan. Note that Inverse Sine is most effective with Intensity settings of 9.9 or lower.

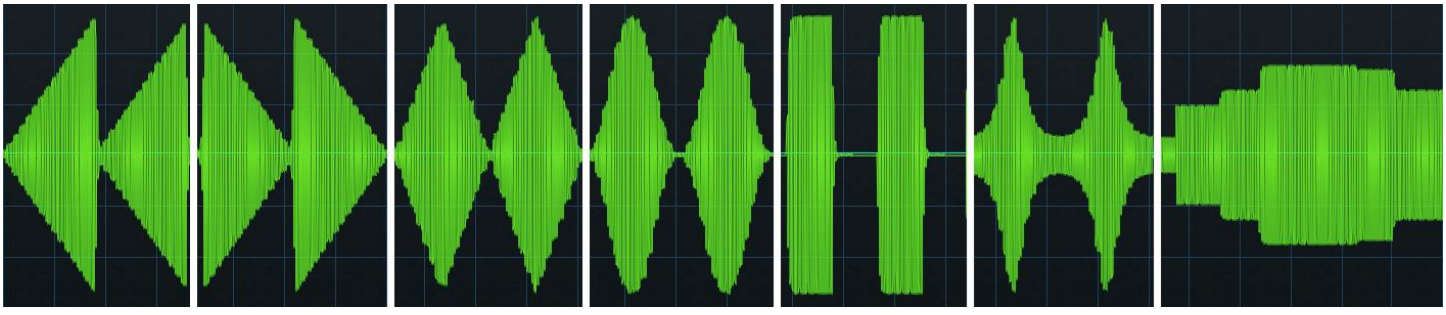
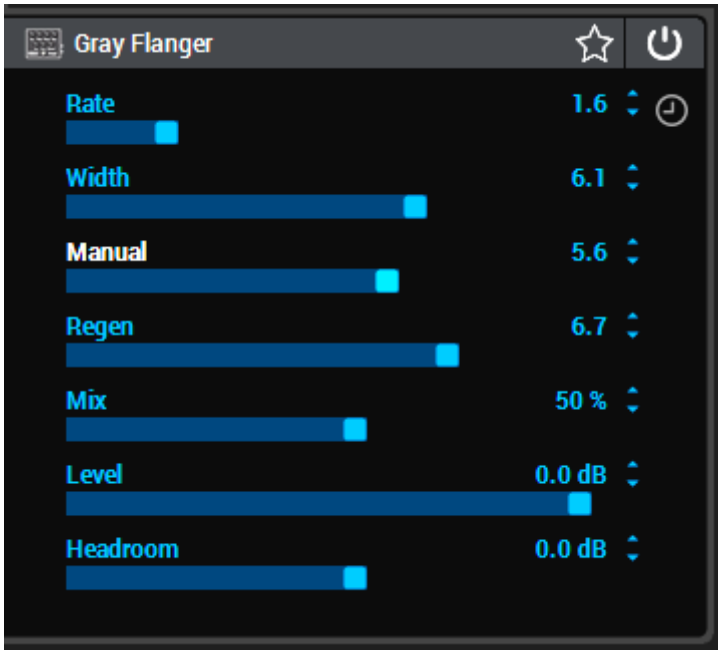


Figure 2.33 Left to right: The LFO's Saw Up, Saw Down, Triangle, Sine, Square, Inverse Sine, and Random wave shapes control the phaser's sweep.

Gray Flanger

The Gray Flanger emulates a popular stompbox flanger and has a limited control set. Most of the controls were covered in the section on Modulation Blocks, except that feedback is called **Regen** in this effect.



A couple of the controls may seem counter-intuitive:

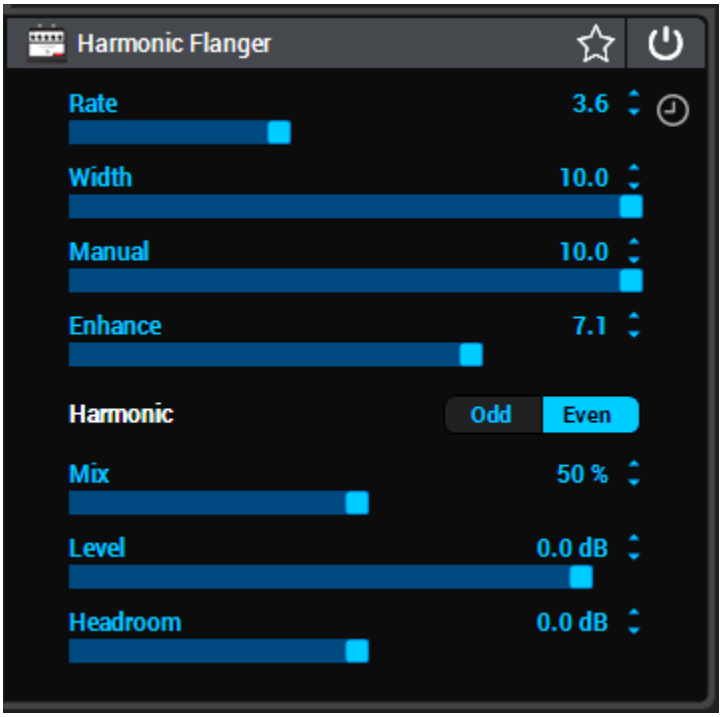
- **Manual** sets the flanging sweep's midpoint. Higher values move the effect to a shorter delay time, while lower values move the effect to a longer delay time.
- **Width** controls the delay variation around the delay set by the Manual control. Higher width values give a wider, more dramatic sweep.

Turning up **Width** and **Manual** all the way covers the widest flanging range. If Width is up all the way, Manual has no effect because the flanger is already covering the widest possible range.

Tip: To control the flanging effect with an Expression Pedal, set Width to 0.0, and assign the pedal to Manual.

Harmonic Flanger

Although the sonic character differs somewhat from the Gray Flanger, the **Rate**, **Width**, **Manual**, **Mix**, **Level**, and **Headroom** controls work similarly. The **Enhance** and **Harmonic** parameters are the Harmonic Flanger's main differentiating features compared to other Helix flangers.



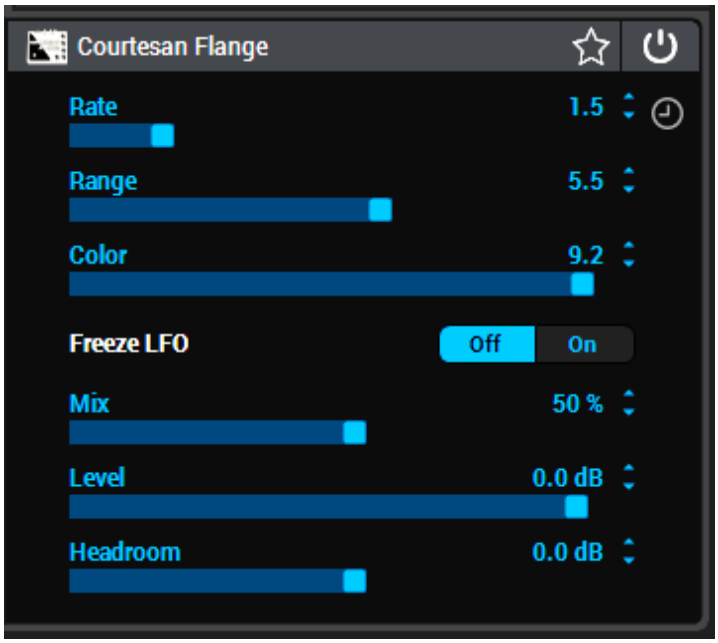
Turning up **Width** and **Manual** all the way covers the widest flanging range. With Width at maximum, Manual has no effect because the flanger is already covering the widest possible range. The Enhance and Harmonic parameters do the following:

- Higher **Enhance** values increase the amount of feedback, and also affect the delay time.
- The **Even** Harmonic setting has a hollower sound, while the **Odd** setting sounds sharper.

Tip: With Enhance at maximum, the Harmonic Flanger will self-oscillate.

Courtesan Flanger

The **Rate**, **Range**, **Mix**, **Level**, and **Headroom** parameters work the same way as the flangers described previously (Range is another name for Initial Delay, as described in the section on Modulation Blocks.) Again, higher Range values increase the initial delay time, which sets the lower range of the flange sweep. The unique feature here is the **Freeze LFO** option (described below).



- **Color** is like the feedback control in other flangers. Higher values give a more resonant sound.
- Setting **Freeze LFO** to On disconnects the LFO. With the LFO disconnected, the Rate parameter does nothing, the effect no longer sweeps, and the sound "freezes" at the current Range setting. The resulting static (i.e., non-sweeping) frequency response is called a *comb filter* (fig. 2.34). Range sets the Comb Filter frequency. Color edits the resonance.

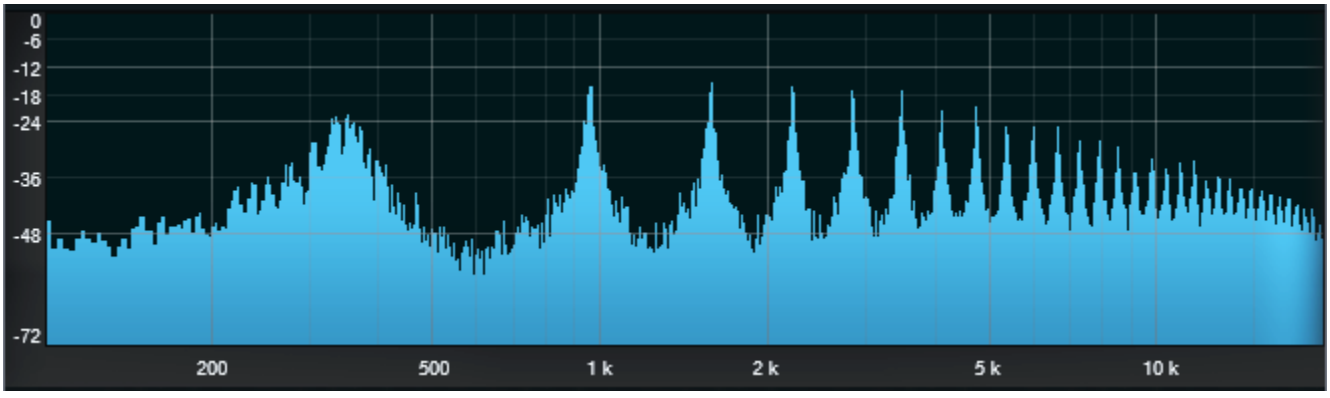


Figure 2.34 The Comb filter gets its name because the frequency response looks like a comb's teeth.

Dynamix Flanger

This flexible flanger has three control options: LFO, Envelope, or Manual. This is a complex effect, particularly with Envelope control, so we'll cover each control option as if it's an individual effect.

LFO Control

Fig. 2.35 shows the controls that affect the LFO mode.

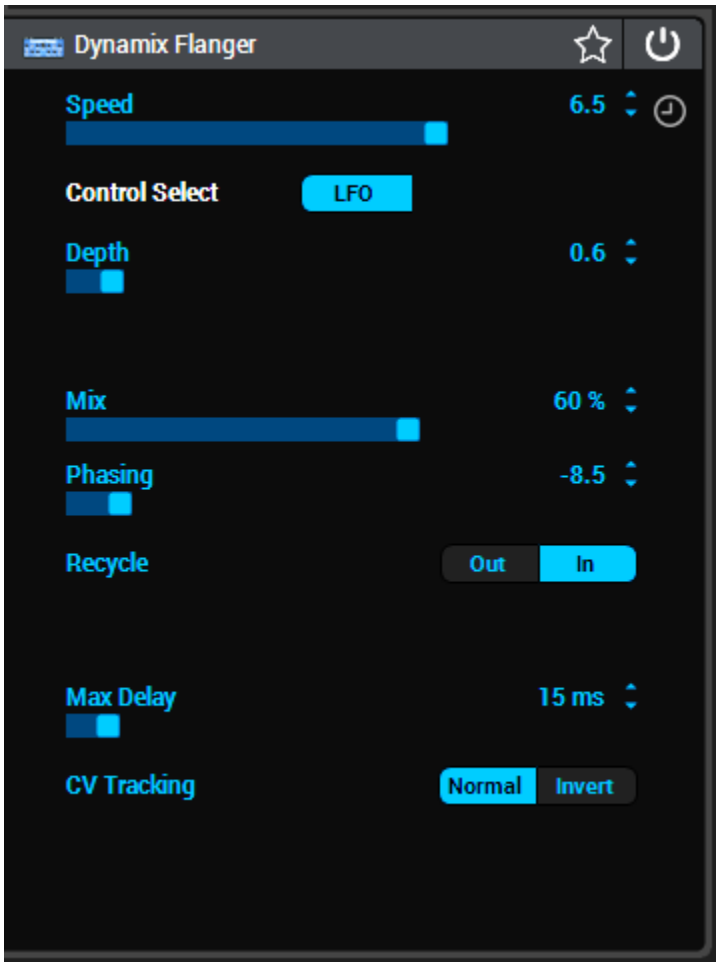


Figure 2.35 The relevant controls for LFO mode.

- **Speed** (Rate), **Depth**, and **Mix** are standard LFO flanger controls.
- **Phasing** chooses two different flanging characters. Negative values produce in-phase flanging, which gives a more resonant sound. Positive values create out-of-phase flanging, for a hollower sound. At 0, there's no flanging effect. Also at this setting, there's no audio with Mix set to 100%.
- **Recycle** adds feedback, which emphasizes the Phasing parameter's effect.
- **Max Delay** sets the longest point of the flanging sweep, up to 100 ms. Chorusing and chorused echoes are available at longer delays, with low Depth settings. If the delay is long enough to give audible echoes, Recycle adds repeats to the echo effect.
- **CV Tracking** changes the sweep direction, so that it spends more time at the top part of the sweep. It's more useful for Envelope control.

Envelope Control

With this control mode (fig. 2.36), the flanger's sweep responds to the dynamics of your playing instead of an LFO. Hitting your string after a brief period of silence triggers (or re-triggers) the sweep. Until the next silence, the sweep tracks your dynamics. **Mix**, **Phasing**, **Recycle**, and **Max Delay** work as described above.

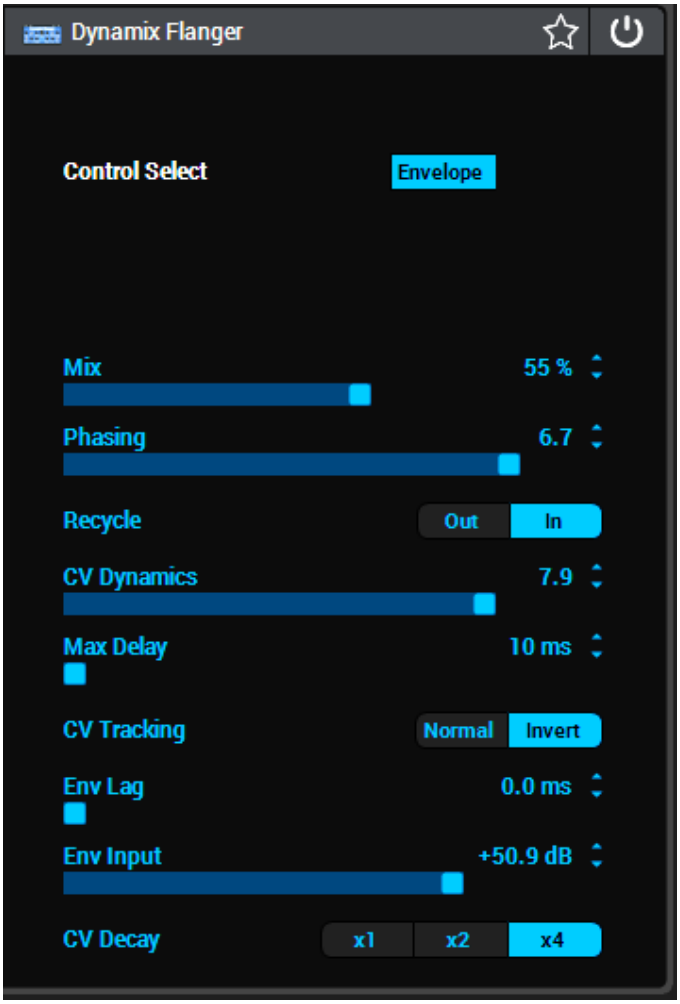


Figure 2.36 The relevant controls for Envelope control mode.

Note that the envelope controls interact with each other. Referring to fig. 2.37, here's how they work:

- **CV Dynamics** sets the sweep range, based on your dynamics. With low settings, playing dynamically covers a narrow sweep range. Higher settings cover a wider sweep range for similar dynamic changes.
- **Env Input** matches your playing's dynamics to the sweep range. Set this so that hitting the strings hardest covers the highest part of the sweep set in CV Dynamics.
- When first triggered, **CV Tracking** determines whether the initial sweep goes from high to low (Normal), or low to high (Invert).
- **CV Decay** sets the initial sweep time, from X1 (barely noticeable) to X4 (still fairly fast, but obvious). It also smooths the envelope, so with chords, the envelope sweeps smoothly instead of jittering.
- **Env Lag** introduces a slight delay before the sweep starts. Note that when re-triggered from silence, the 0.0 ms setting seems to be the same duration as the 100 ms setting. For minimum Env Lag, choose a 0.1 ms setting.

Fig. 2.37 shows the relationship of these parameters for a decaying guitar chord with maximum Env Lag, and a long CV Decay.

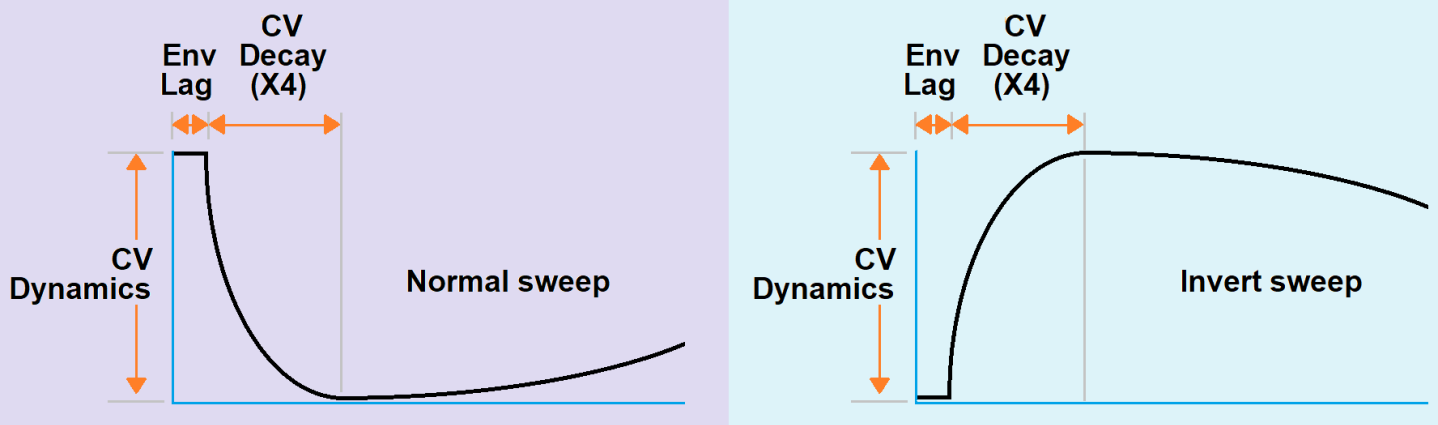


Figure 2.37 How the CV Dynamics, Env Lag, and CV Decay parameters interact.

Manual Control

With manual control (fig. 2.38), no parameters are affected by the LFO or the Envelope. The remaining parameters work as described for LFO and Envelope control, and they are good candidates for use with the Expression Pedal.

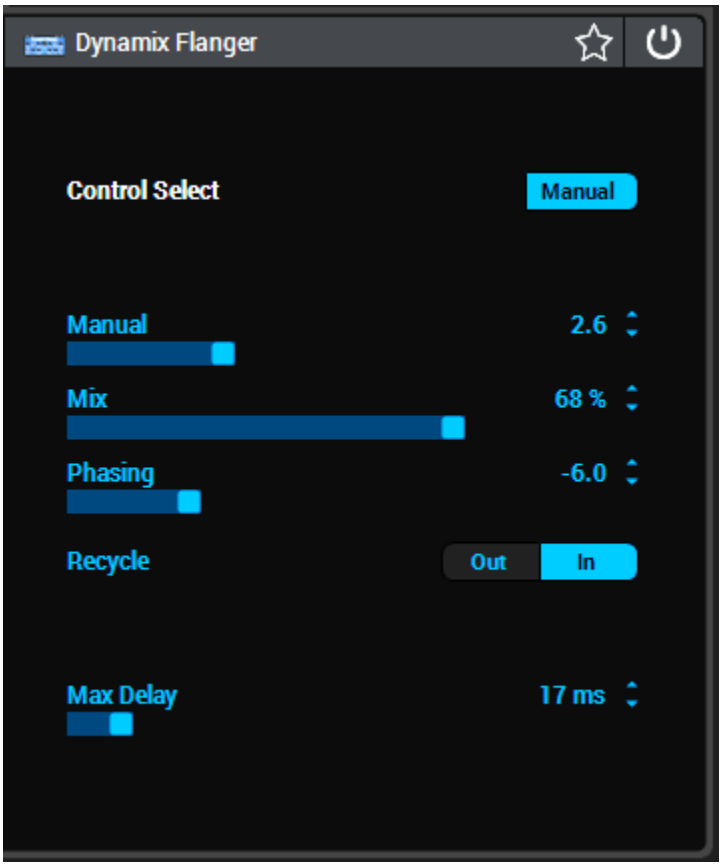


Figure 2.38 Parameters that are relevant to Manual control.

Chorus

Typical chorus effects thicken an instrument's sound, but this Chorus also provides an ADT (Automatic Double-Tracking) effect. The screenshot shows parameter values for a typical ADT effect. The Predelay parameter, which isn't common on many chorus effects, makes this possible. I prefer using this effect in stereo, but it works in mono too.

Chorusing is a parallel effect that mixes a dry signal with a delayed, modulated signal. The controls are:

- **Speed** controls the rate of change. Slower speeds are more ethereal, faster ones are more "bubbly."
- **Depth** affects how modulation affects the delay time. More depth gives a more intense effect.
- **Wave Shape** offers the usual Line 6 Original wave shapes for modulation (fig. 2.39). Note that Inverse Sine is most effective with Intensity settings of 9.9 or lower.

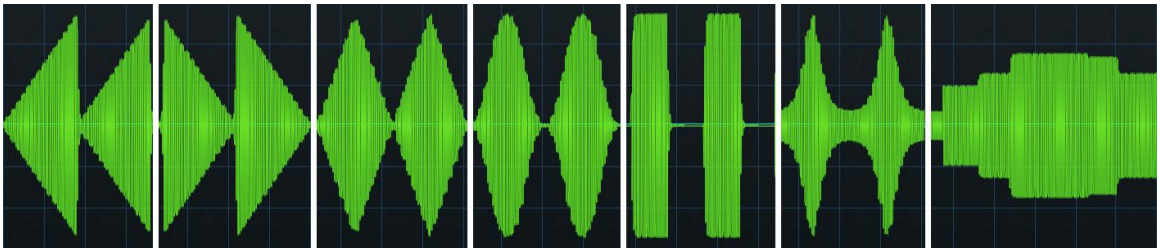


Figure 2.39 Left to right: Saw Up, Saw Down, Triangle, Sine, Square, Inverse Sine, and Random.

- Higher **Tone** settings sound brighter, lower **Tone** settings sound warmer.
- **Spread** appears only in the stereo version. Higher values widen the stereo image.
- **Mix** chooses percentage of delayed to dry sound.
- **Level** adjusts the output level.
- Higher **Predelay** values (longer delays) provide convincing ADT effects. Short delays sound more like complex flanging. Medium delays occupy the sonic space between chorus and delay. **Rate**, **Depth**, and **Mix** alter the sound of the "double-tracked" audio. For the ADT effect, I prefer using a sine or triangle Wave Shape.

70s Chorus

The chorus on which this is modeled was known for Chorus *and* Vibrato effects. We'll look at both (fig. 2.40).

Figure 2.40 Left, typical Chorus settings. Right, typical Vibrato settings.

Rate, **Depth**, **Spread**, and **Level** are parameters we've covered before. For information on Headroom, see Modulation Blocks. In the stereo version, the Stereo parameter is unique to the 70s Chorus:

- **True** provides stereo imaging for the chorused audio.
- **Classic** sends the dry audio to the left channel, and the delayed/modulated audio to the right channel. When combined in mono, they sound like chorusing.

Other tips:

- For vibrato with the stereo effect version, keep Spread at minimum. Otherwise, the effect won't collapse well to mono.
- For vibrato with the mono or stereo version, set Mix to 100%. Other settings sound more like chorusing. This is true for both the True and Classic stereo options.
- If you assign a footswitch to switch between Chorus and Vibrato, also assign Spread and Mix to the same switch. When using the stereo mode, the following settings collapse well to mono:
    - Spread = 10.0 for Chorus.
    - Mix = 50% for Chorus, 100% for Vibrato.

PlastiChorus

This model is based on a chorus known for its sweet chorus sound. It had a switch to choose between direct and stereo outputs, but a popular mod changed this switch to select chorus or vibrato. Fig. 2.41 shows typical settings for Chorus and Vibrato.

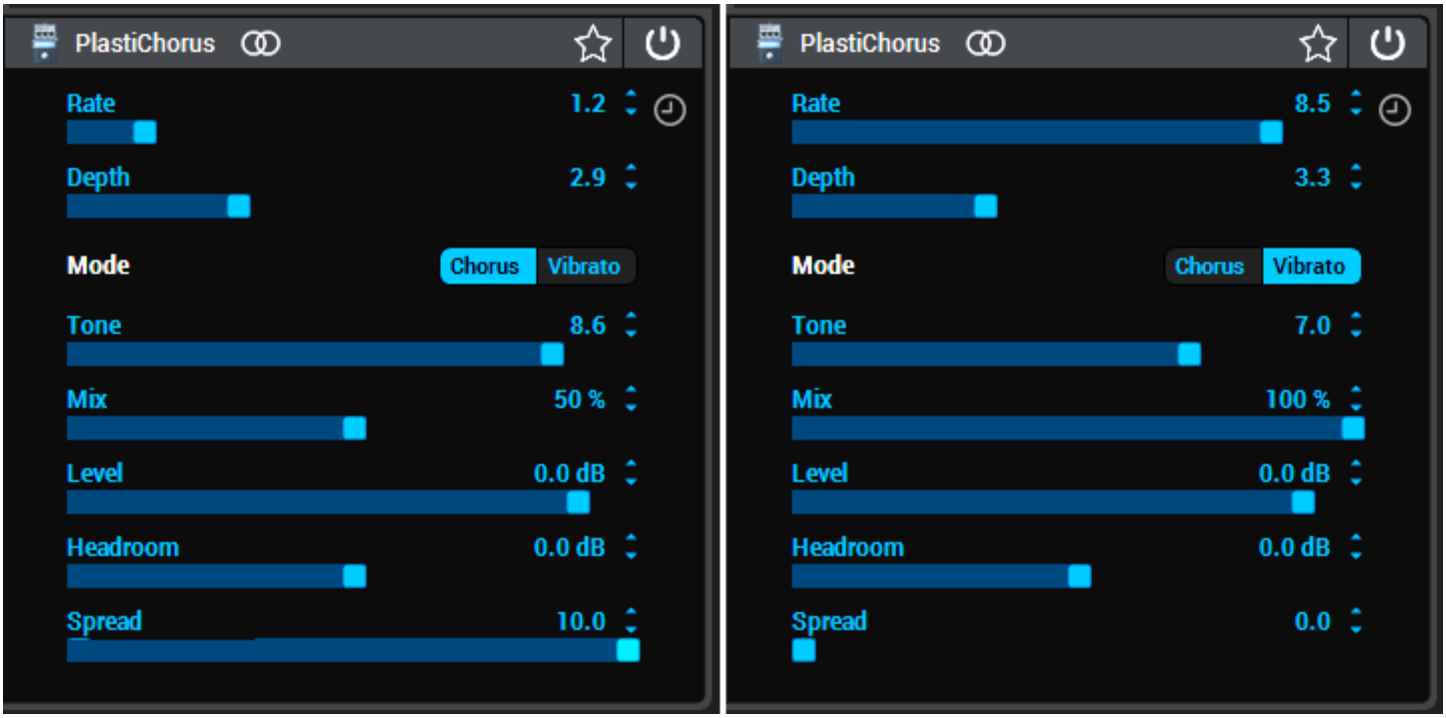


Figure 2.41 Typical PlastiChorus Chorus (left) and Vibrato (right) settings.

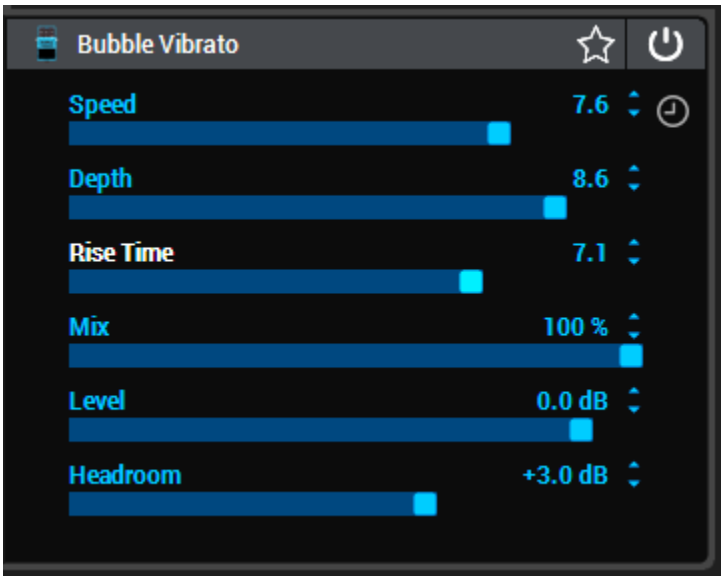
We've already covered all but one of these controls in the section Modulation Blocks. The **Tone** control boosts the highs at 10.0. It reduces the highs at 0.0. It also gives a very slight lower midrange lift.

As with most effects that provide both chorusing and vibrato:

- For stereo chorusing, set Spread at maximum for the widest stereo image.
- For stereo or mono chorusing, set Mix to 50%.
- For stereo vibrato, set Spread to minimum.
- For stereo or mono vibrato, set Mix to 100%.

Bubble Vibrato

This nails the sound of bucket-brigade technology vibrato. An HX Stomp costs the same as the current cost of the original hardware. So, if you want this sound, the Bubble Vibrato alone justifies the HX Stomp's existence.



All the controls have been covered before (the stereo version also includes a Spread parameter) except for **Rise Time**, which is the Bubble Vibrato's standout feature. When you enable the effect, Rise Time fades in the vibrato. Higher values lengthen the fade-in time.

For more control over vibrato, assign the **Depth** to an Expression Pedal. However, note that Rise Time also affects Depth:

- If you slam the pedal from full off to full on, Rise Time causes the effect to fade in after the pedal is all the way down.
- If you slam the pedal back to full off, Rise Time becomes a decay time and fades the vibrato out after the pedal is all the way back. Having these two options available is great for live performance.

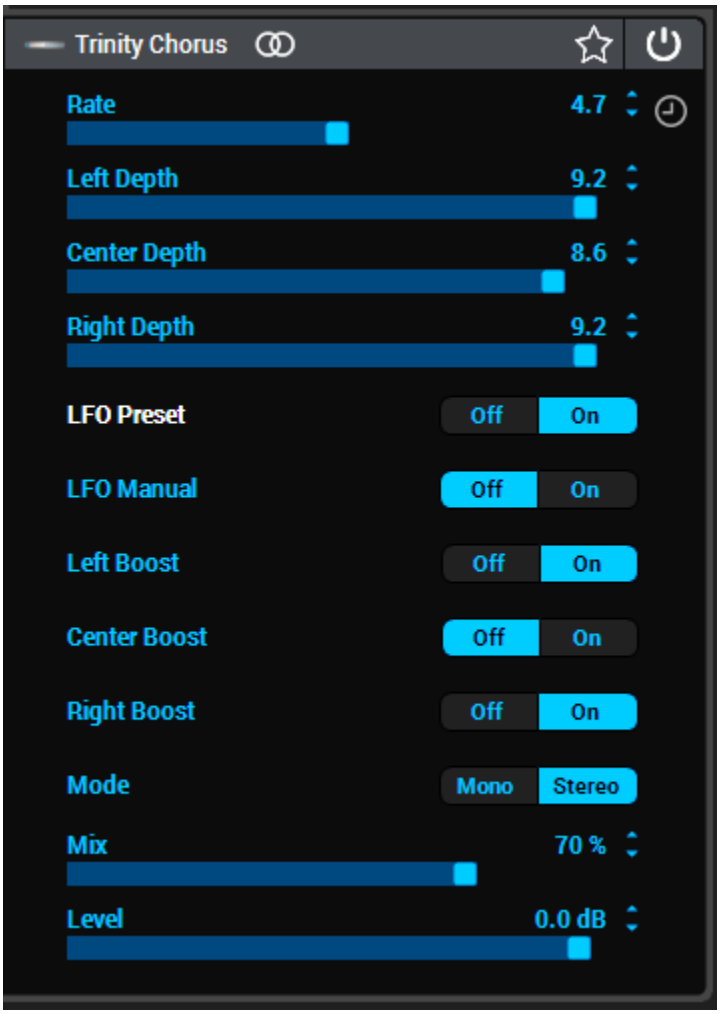
Like the original, a gentle high-frequency rolloff starts around 400 Hz. Also note this pedal saturates nicely at lower frequencies.

Tech Talk: Bucket-Brigade Device (BBD)

Before digital technology became common and affordable, early delay-based devices used bucket-brigade technology, which were superior to an even earlier analog technology based on the CCD (Charge-Coupled Device). BBDs have a string of capacitors in series, with the analog signal represented by an electrical charge that's shuttled from one capacitor to the next. This is like the way firefighters used to pass water from one to another in the process of putting out a fire, hence the "bucket brigade" name. Using analog technology to create delay required tradeoffs that affected distortion, level, and high-frequency response. These design decisions gave BBD-based devices different sounds, as well as a different sonic character compared to digital delays.

Trinity Chorus

This is available only as a stereo effect (although a switch in the effect can collapse the output to mono). Its unique feature is having three different BBD delays, for the left, center, and right channels. Each delay has individual **Depth** and **Boost** parameters, as well as its own LFO. Therefore, the Trinity Chorus can create complex, swirling chorus effects.



The Depth controls work as expected, but how to set the LFO Rate is not obvious. Here's how it works:

- **LFO Preset On, LFO Manual Off.** Think of this as "plug and play" mode. Rate and Depth parameters have no effect. The Rate also ignores Note Sync. The three delay lines have fixed LFO speeds optimized to give shimmering, lush chorus effects.
- **LFO Preset Off, LFO Manual Off.** The Rate control has no effect and there's no modulation. Changing the Depth controls is like freezing the LFO at different places along the LFO waveform.
- **LFO Preset Off, LFO Manual On.** Rate varies the LFO speed, from slow chorusing, to fast, vibrato-like effects.
- **LFO Preset On, LFO Manual On.** The Preset LFO is enabled, but you can modulate the rate further by varying the Rate parameter.

Tip: With Helix Native used in a DAW, and both LFO Preset and LFO Manual off, using automation to vary the three Depth controls gives another type of chorus effect.

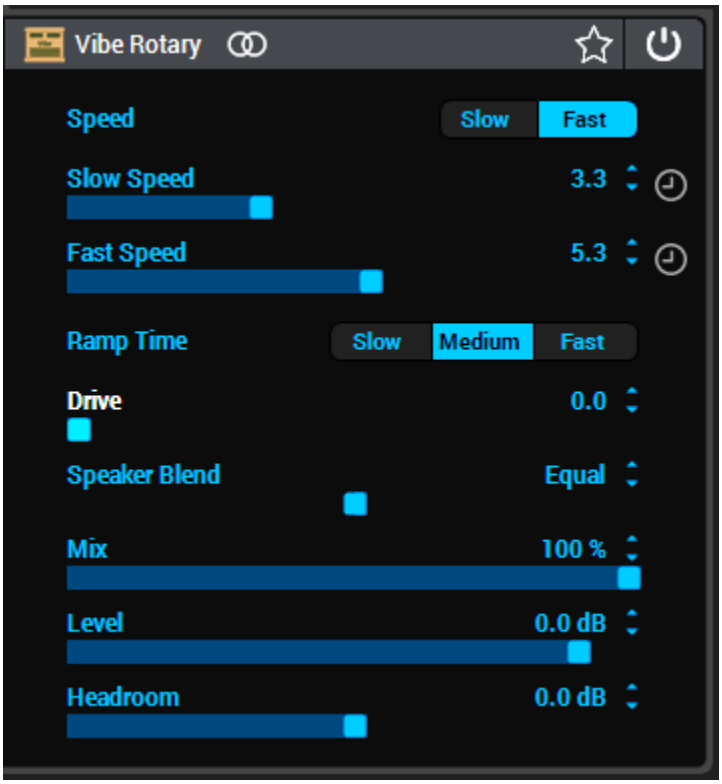
The **Boost** parameters increase the apparent amount of delayed signal in the mix, by adding a slight amount of feedback.

Like most delay-based effects, **Mix** varies the ratio of processed to dry sound. However, with this effect it's not crucial to have the Mix at 50% for the most intense sound. With Mix at 100%, the three delays are still working against each other to create complex time-delay sounds. Mix is more about setting how much chorus you want in the effect, rather than the chorus intensity.

The Trinity Chorus effect collapses well to mono, but you can ensure a mono output with the **Mono/Stereo** mode switch. In Mono, a center-channel buildup increases the output level compared to the stereo setting. For more about center-channel buildup, see the related Tech Talk.

Rotary Speaker Effects: Vibe, 122, 145, and Triple

These effects emulate the sound of mechanical rotating speakers. They are available only in stereo. The Vibe, 122, and 145 effects have the same parameters. However, Line 6 has taken some liberties with the Vibe Rotary.



The Vibe Rotary defaults to 0 **Drive** because the unit on which this is based did not have an amplifier. Creating a drive effect required an external amp (e.g., a guitar amp) or effect, which this parameter emulates. Also, the original device didn't have separate high- and low-frequency drivers, only a single 10" speaker. *Personal bias alert:* I'm not a purist—as long as you can obtain the original sound, I don't mind "bonus" features. If you want the most authentic Vibe Rotary sound, set the **Speaker Blend** parameter to Equal.

Note that the main tonal difference between the 122 Rotary and 145 Rotary comes from emulating their two different cabinet sizes. The 122's larger cabinet gives a fuller tone.

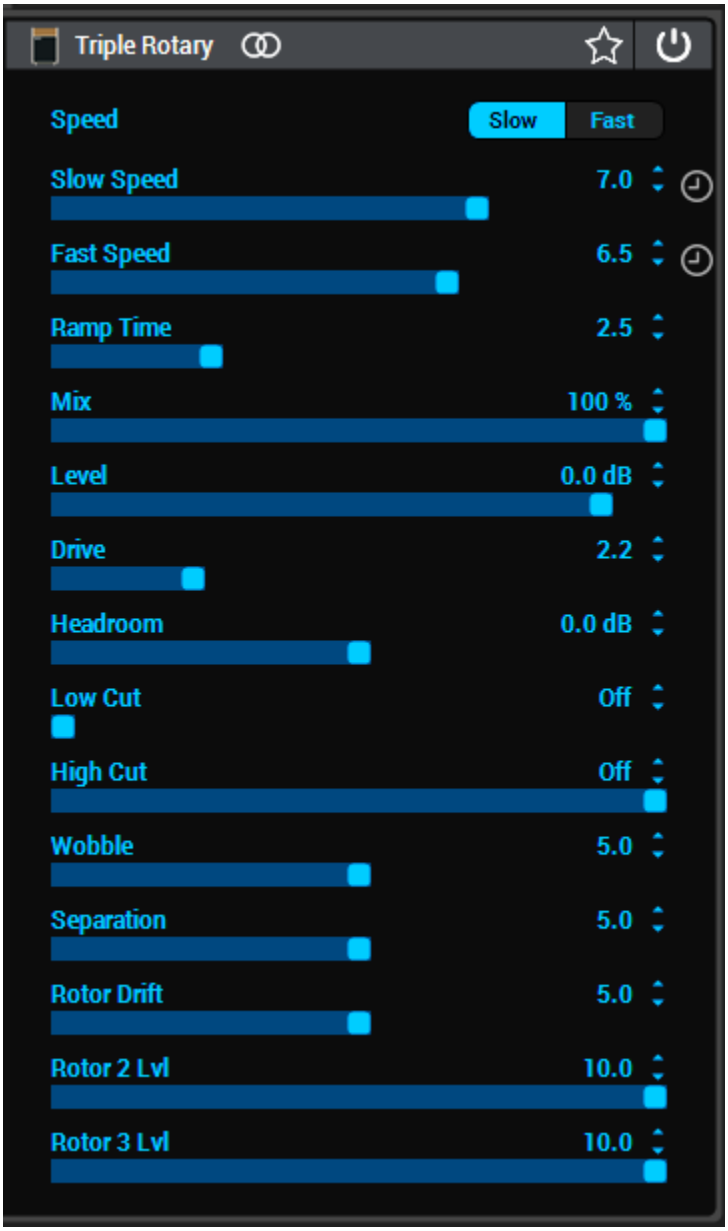
Much of the distinctive rotating speaker sound comes from being able to change the rotation **Speed** from Slow to Fast. These mechanical devices had inertia, so the change wasn't instantaneous. In the virtual world, we have options. Here's how the speed-related parameters work:

- **Speed** has two options, Slow and Fast. To emulate a physical rotating speaker's functionality, you'd assign this parameter to a footswitch, and toggle between the two settings.
- **Slow Speed** sets the speaker rotation speed for the Slow mode.
- **Fast Speed** sets the speaker rotation speed for the Fast mode.
- **Ramp Time** determines the transition time when switching between the slow and fast modes. The Slow setting acts like a physical rotating speaker. Fast and Medium exist only in the virtual world.
- **Drive** emulates overdriving the speaker's internal amp. This was a common technique for rock groups who wanted a grittier organ sound.

- **Speaker Blend** selects the ratio of the low-frequency woofer to the high-frequency horn speaker. In a mix, you might want to tilt more to the Horn to cut through the mix better. This can also avoid having the low end add to any existing lower midrange "mud."
- **Mix**, **Level**, and **Headroom** work as described previously in Modulation Blocks.

Tip: For a novel chorusing-type sound in the studio with Helix Native, send your guitar track to two buses. Insert the same rotating speaker emulator preset in both buses, using the Slow speed, and Mix set to 100%. Set one emulator for Horn only, and the other for Woofer only. Pan them slightly opposite to each other (e.g., center-left and center-right). This produces a wide, gorgeous stereo spread. To reduce the width, rather than change the panning, move Mix more toward 50% so both buses have some dry signal in common.

Triple Rotary



Sound Designer Sam Hwang's comments in the release notes explain clearly what makes the Triple Rotary different from other rotary speakers. The following is lightly edited: "The Yamaha RA-200 Rotary speaker

inspired the Line 6 Triple Rotary. The original RA-200 was a combo solid-state amplifier, designed to be used with organs much like other rotary speakers. However, traditional rotary speakers typically have rotating horns and a rotating drum over a woofer. The Yamaha designs had midrange speakers which rotated vertically on top of a traditional non-rotating speaker cabinet."

Sam notes the goal was a versatile stereo effect, so it's not limited to a specific cab. This allows choosing a cab that complements the effect for your specific needs. *Personal bias alert:* I generally prefer the various Fender cabs (US Princess, US Deluxe, Double C12N, Tweed P10R) with the Triple Rotary, but all cabs provide their own unique effects.

Regarding the parameters, **Speed**, **Slow Speed**, **Fast Speed**, **Ramp Time**, and **Drive** work as described above. **Mix**, **Level**, and **Headroom** work as described previously in Modulation Blocks. The Triple Rotary's unique parameters are:

- **Low Cut** reduces the bass response by trimming low frequencies from 20 Hz up to 500 Hz. The Off setting bypasses the Low Cut filter.
- **High Cut** reduces the treble response by trimming high frequencies from 500 Hz to 20 kHz. The Off setting bypasses the High Cut filter.
- When set to 0, **Wobble** emulates a mechanically balanced rotating speaker setup. Increasing Wobble introduces more eccentricity in the speaker's rotation. This emulates the effect of more wear and tear.
- **Separation** narrows or widens the rotating speaker effect.
- **Rotor Drift** emulates how belt-driven speaker rotation changed over time, due to the aging of the belts. This caused the three speakers to be less in sync with each other. The result was subtle modulation effects among the three rotating speakers.
- **Rotor 2 Lvl** adjusts the individual level of the second rotor.
- **Rotor 3 Lvl** sets the individual volume of the third rotor.

Wild Lezlies Preset

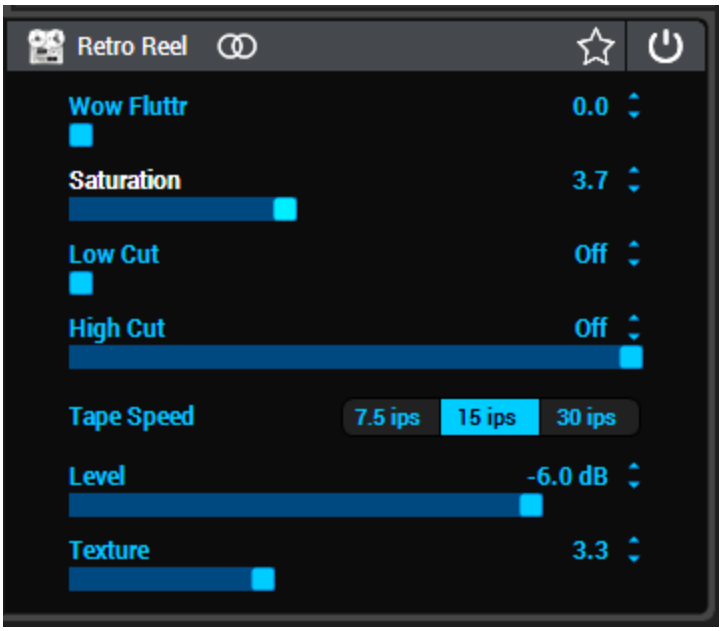
The preset **Wild Lezlies** in the *Free Files > Chapter 2 Presets* folder highlights a novel application for rotating speaker emulators. Insert two speaker emulator processors in parallel, and set the parameters as follows:

- Set one's **Slow Speed** to 0.0.
- Set the other's **Slow Speed** to around 7.0 or so.
- Choose 100% **Mix** for one, and **50%** Mix for the other.
- At the Mixer, select **Inverted** for the B Polarity instead of Normal. Keep **Pan** centered for both paths, which should also have the same **Level**.

This produces a wild stereo effect that "tosses and turns" more than conventional chorusing. For even wilder psycho-acoustic panning effects, set both Mix controls to 100%.

Retro Reel

Reality check: There is no single "tape emulation" sound. How tape affected audio depended on the tape formulation, the input level, how well the recorder was maintained, and how an engineer *biased* tape. (You couldn't have low distortion, low noise, *and* excellent high-frequency response—the way you applied a supersonic bias signal to the tape prioritized the characteristic you wanted.) So, Retro Reel is not about a *particular* tape sound, but one that's a "greatest hits" of tape sounds in general.



The reason why people want to emulate tape, even in the digital age, is because tape is a signal processor: It saturates the audio, while adding a low-frequency "bump" that brings up kick and lower bass notes. These two factors can give a track more punch. However, tape also has issues like wow, flutter, and frequency response anomalies, which you may not want to include. Here are the parameters:

- **Wow Fluttr** imposes tape speed variations. *Personal bias alert:* When digital recording arrived, I was thrilled I'd never have to deal with speed variations again. So, my parameter is almost always set to 0.0.
- **Saturation** distorts the signal and adds harmonics. Higher Saturation settings or input levels increase saturation. See Tech Talk: About Saturation for more information.
- **Low Cut** reduces levels at lower frequencies, from 20 Hz up to 1 kHz. Settings below 20 Hz turn off the Low Cut filter.
- **High Cut** reduces levels at higher frequencies, from 20 kHz down to 500 Hz. Settings above 20 kHz turn off the High Cut filter. Repeated tape playback caused a gradual deterioration in the high frequencies, which this control can emulate.
- **Tape Speed** alters the frequency response (fig. 2.42), and the wow and flutter's modulation speed. Note the subtle bass boost in the 15 and 30 ips speeds, at 50 Hz and 100 Hz respectively. Even though 30 ips was better than 15 ips in several ways (e.g., better high-frequency response), many engineers preferred 15 ips. Tape costs were lower, and the 50 Hz boost brought out kick and bass more. This boost was called a *head bump* because it resulted from the physics of tape heads.
- **Texture** replicates the influence of the NAB EQ curve on tape. At higher Saturation levels, higher Texture values give a "woolier," less bright timbre.
- **Level** does what you'd expect.

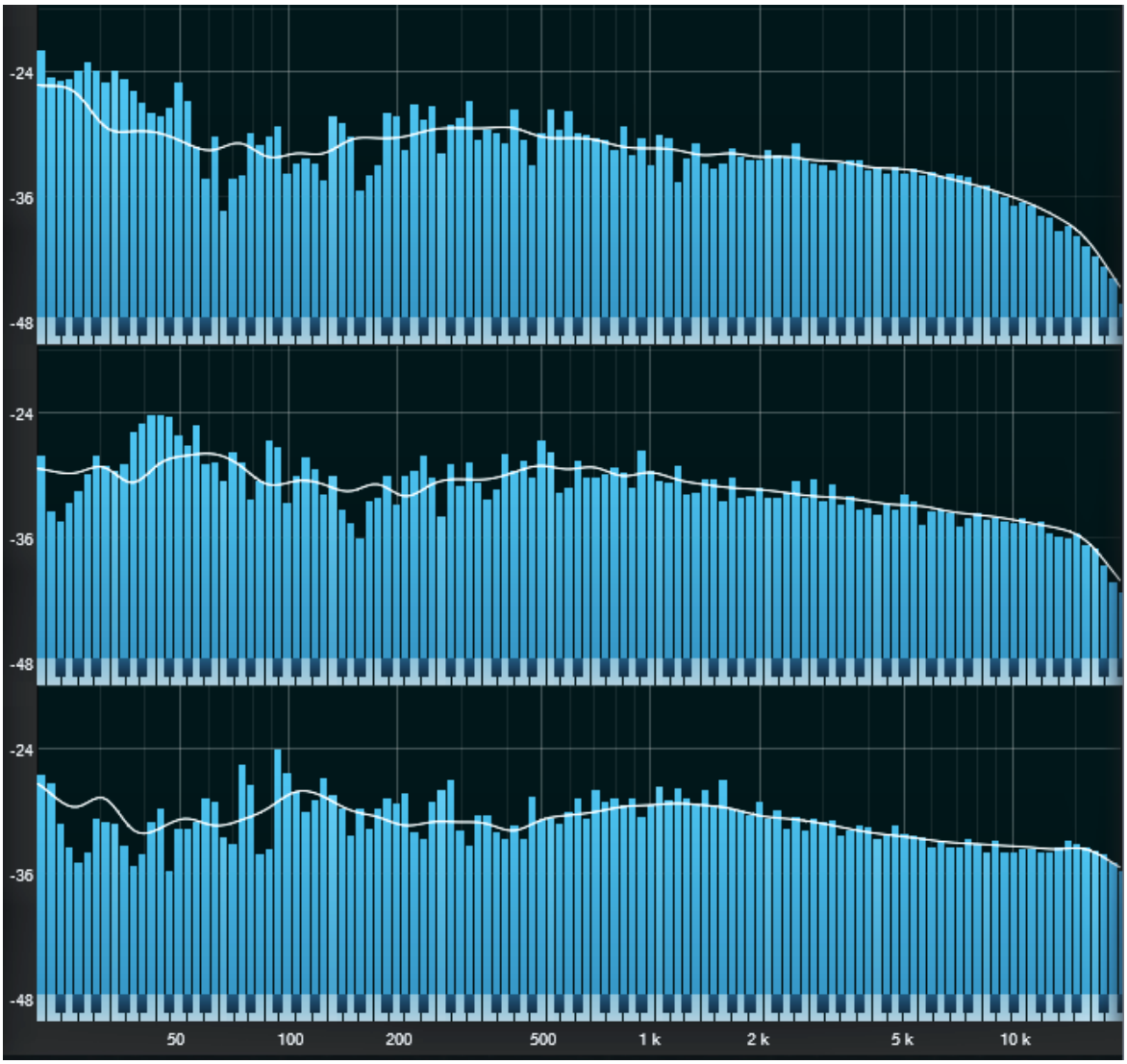


Figure 2.42 Top to bottom: Frequency response at 7.5 ips, 15 ips, and 30 ips. 7.5 ips has the poorest high-frequency response, 30 ips has the best. Also notice the slight midrange dips around 100 Hz, 200 Hz, and 400 Hz for the 7.5, 15, and 30 ips speeds, respectively.

Tech Talk: About Saturation

Fig. 2.43 (next page) shows the harmonics generated with the Saturation parameter setting at 3.7, for different levels of a 500 Hz sine wave tone. Sine waves have no harmonics, so any harmonics in the image are generated solely by the Retro Reel effect.

With a -20 dB input signal, there are no noticeable harmonics. At -17 dB, second- and third-order harmonics start to appear. The sound isn't that different from the -20 dB input signal, yet there's a subtle change—this is the kind of "fairy dust" that people like about tape emulation. At -10 dB, the harmonics all have higher levels, there are additional higher-frequency harmonics, and the saturation is clearly audible.

Saturation reduces dynamic range, which is one reason why tape can add "punch." Another reason is generating harmonics that add brightness. In addition to using saturation to shave the peaks from guitar signals, it's also effective with drums and other instruments.

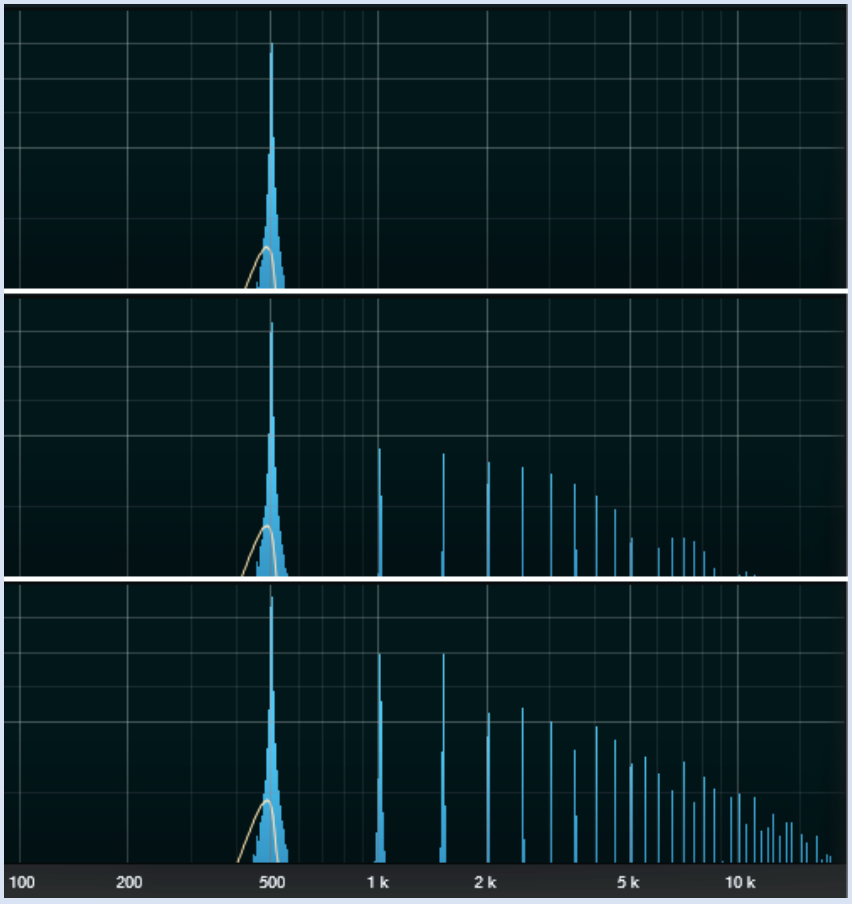


Figure 2.43 Top to bottom: Harmonics produced with input levels of -20, -17, and -10 dB.

Double Take

This optimizes the popular ADT (automatic double-tracking) effect for guitar. Double-tracking involves overdubbing the same part. Because it's impossible for humans to play the *exact* same part twice, there will be slight timing and pitch differences. These differentiate the overdubbed part from the original part. The resulting difference can thicken the sound, or provide two versions of the track to create a stereo image.

The parameters in the screenshot above give a traditional double-tracked sound. Setting Doubles to 2 gives a triple-tracked effect.

Double Take has mono and stereo versions, but benefits the most from stereo:

- **Source** is only in the stereo version (all other parameters are in both the stereo and mono versions. Source chooses how the Double Take will process a stereo input. **Mono** sums the left and right channels. **Stereo** preserves the stereo path through the effect. **Left** processes only the left channel. **Right** processes only the right channel.
- **Doubles** sets the number of voices (overdubs) that are added, and their stereo position:
  - 1: Left output dry, overdub in right.
  - 2: Center dry, overdubs in left and right
  - 3: Left output dry, one overdub left, two overdubs right
  - 4: Center dry, two overdubs left, two overdubs right
- **Slop** edits the doubling's tightness. 0.0 has the least variation, 10.0 the most. However, this also responds to dynamics, as set by the Sensitivity control.
- Higher **Sensitivity** settings cause more Slop when you play harder.
- **Dry** sets the dry level, but the dry signal's stereo location depends on the Doubles setting. With the Source set to stereo, all the right channel input does is generate an overdub. The unprocessed version of this channel doesn't pass through to the output.
- **Wet** balances the overdubs with the dry audio.

My preference is using Double Take for single-note solos, because the responsiveness to dynamics can give double-tracked solos more "feel."

It’s a Through-Zero Flanger, Too!

In stereo, the variations in opposite channels sound like double-tracking. In mono, the effect is more like chorusing, but modifying the mono effect can give wicked *through-zero* flanging (fig. 2.44). See the Tech Talk on Through-Zero Flanging for more information about this signal processing technique.

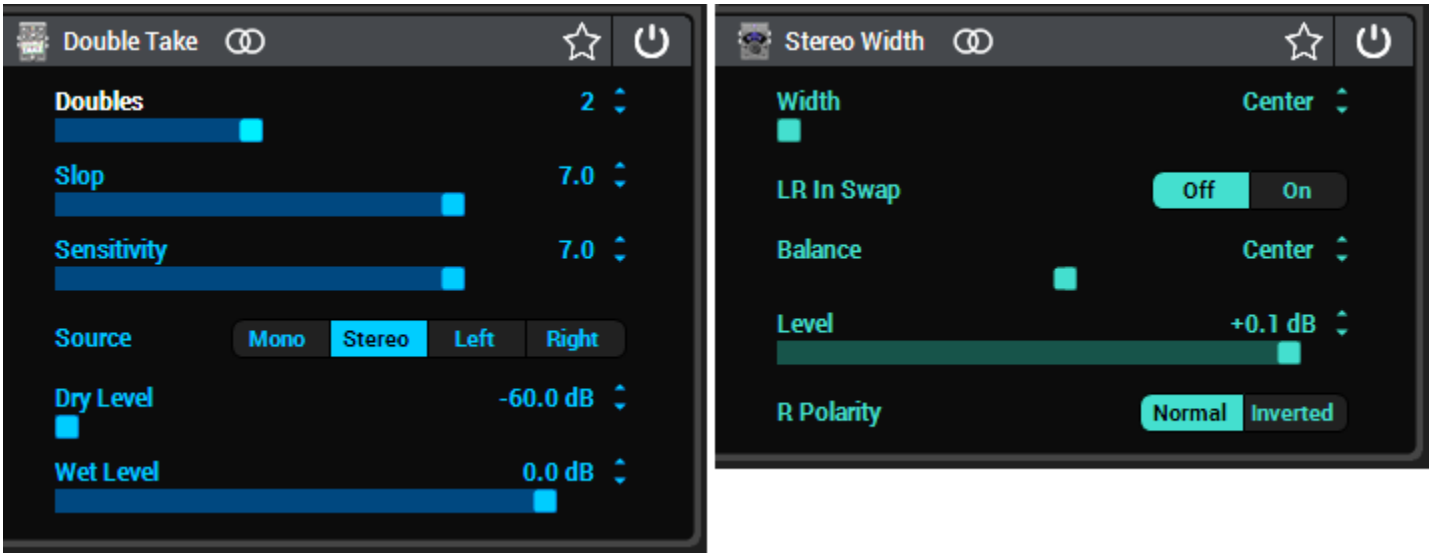


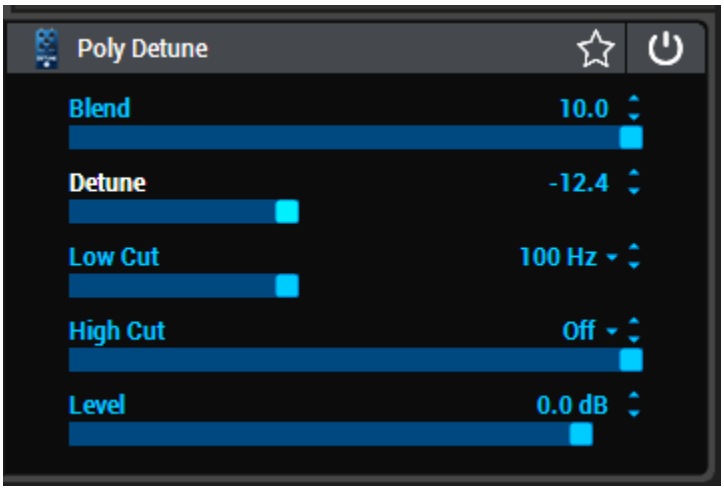
Figure 2.44 The Stereo Width block can narrow the Double Take’s stereo width to mono.

Set the Double Take’s Dry Level to minimum and Doubles to 2. This places two varying signals in the left and right channels. Summing them to mono with the Stereo Width block takes advantage of one channel being able to sometimes get ahead of the other one in time. Therefore, the audio passes over the through-zero point. Put an Amp+Cab in front of this, play some thick chords, and enjoy the flanging.

Poly Detune

This mono-only effect soaks up processing power like a sponge, but can give beautiful, shimmering effects that provide an alternative to conventional chorusing. The Poly Detune effect has a bit more clarity and less diffusion than traditional choruses. The **Detune** parameter sets the chorusing intensity.

Because this effect is mono, unlike some chorusing effects it collapses perfectly to mono.



Note that **Blend** always mixes in the dry sound. If the output could be set for detuned sound only, then it would be possible to do techniques like tune your guitar to a track that's slightly off-pitch (e.g., tune to a track with a slightly out-of-tune piano). For a workaround on how to do this, see the section on the Poly Pitch.

Ring Modulator

This lo-fi effect produces the sum and difference of two input frequencies. The **Frequency** parameter is called the *carrier,* and your guitar is the *modulator.* For example, if the guitar is playing A 440 and the Frequency is set to 300 Hz, you'll hear two tones—140 Hz and 740 Hz.

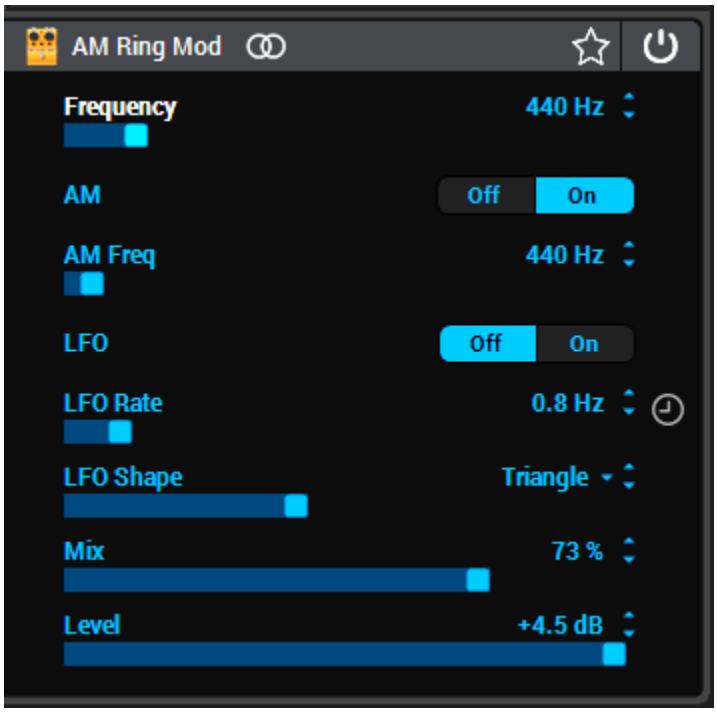
The **AM** (Amplitude Modulation) function adds further modulation. Tremolo is an example of low-frequency AM, but the **AM Freq** parameter included here uses high frequencies to generate additional harmonics.

The **LFO** modulates the AM frequency, which makes the sound even crazier. The Random LFO Shape isn't included in this effect, which I feel is a missed opportunity.

Musical Applications

Most musicians wonder how to use a ring modulator musically. Here are some examples:

- Set both the Frequency and AM Freq to the pitch of the key you're using. With the settings shown in the screenshot, which are optimized for the key of A, play a barre chord A, then do a G and D. E doesn't work as well. The sound will be mostly harmonic and have a cool kind of roughness. This sound is useful for more than just special effects.

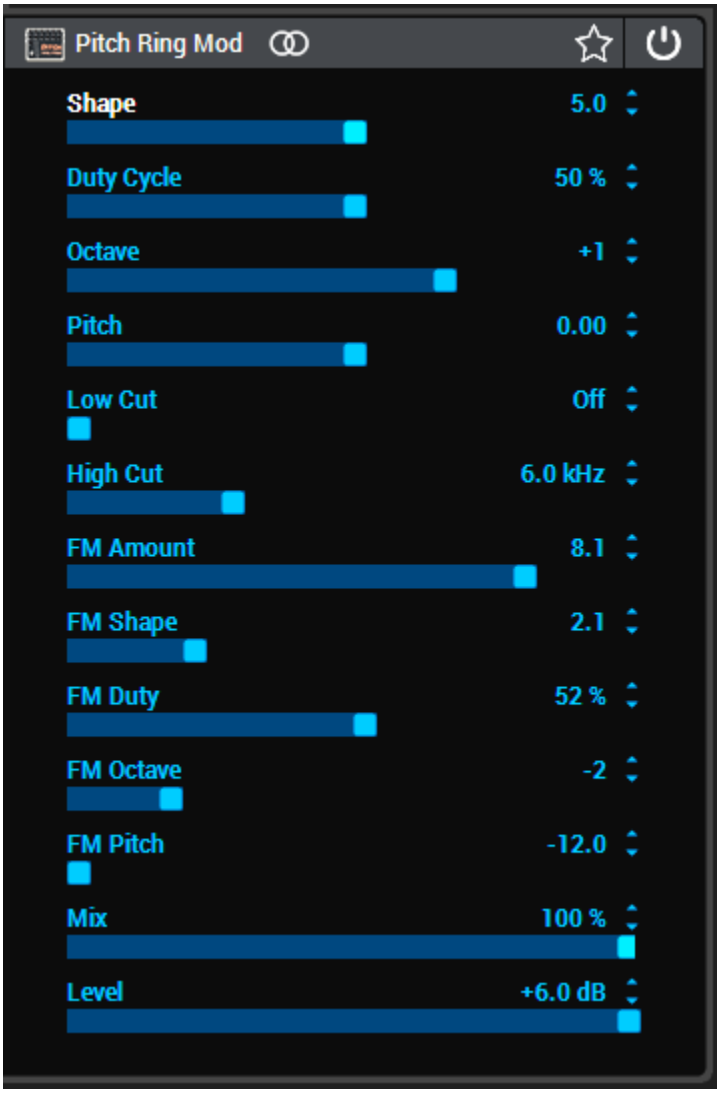


Also try these other tips:

- Turn the Mix down to add some background "fur" to your dry sound.
- Try light ring modulation on chord stabs and muted strums.
- Ring modulation can sound amazing with drums, especially individual drum sounds (e.g., toms).

Pitch Ring Modulator

If *Mad Max: Fury Road* was a signal processor, this would be it—cacophony in a box. I'm not even going to attempt to explain what the controls do, but they involve frequency modulation, pitch shifting, and waveform mangling techniques that may be illegal in some countries. Note that this is available only as a stereo effect, but it collapses perfectly to mono.



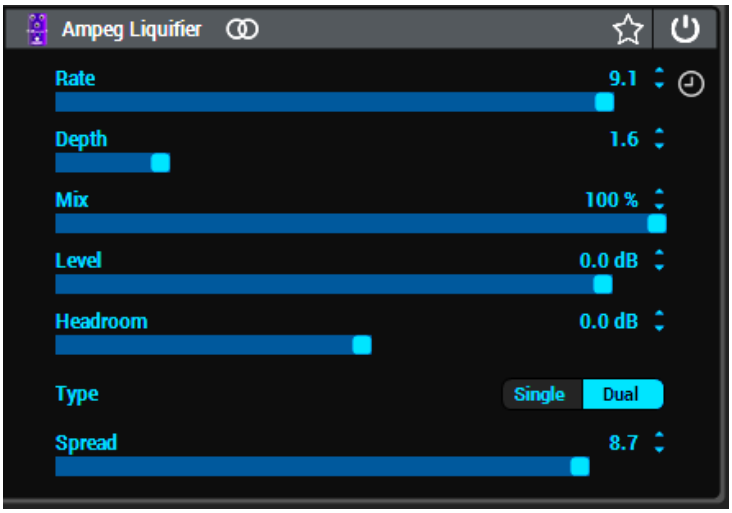
Your mission, should you decide to accept it, is to try and figure this thing out. To familiarize yourself with the Pitch Ring Modulator, turn down **FM Amount** all the way, leave **Low Cut** and **High Cut** off, set **Mix** to 100%, and play around with the top four parameters to hear what they do. Then, turn up **FM Amount**, and edit the various FM parameters.

Although I've emphasized the bizarre aspects because this effect *is* bizarre, there are musical uses. One of my favorites is with palm muted strings when you want to add non-melodic, percussive effects, and with sound sources like electronic drums.

Another good use is to find some parameters that do shocking things when you move an Expression Pedal. Then, use the resulting sound to scare people.

Ampeg Liquifier

The Ampeg Liquifier chorus was designed for bass as well as guitar. Its novel feature is having two chorus paths, which lag and lead each other as the delay time changes. This combines the concept of through-zero flanging with chorusing, because the chorused signal cancels briefly (albeit subtly) when the two chorus paths cross over each other.



The **Rate**, **Depth**, **Mix**, **Level**, **Headroom**, and **Spread** parameters are conventional, and work as described in the Modulation Blocks section. However, unlike the physical pedal on which this effect is modeled, there's a **Single/Dual** switch to select between one or two chorus paths.

Although this may *look* like a simple chorus, it can obtain unique effects.

- **Chorus (thickest sound).** Use the Dual Type and set Mix to 50%. At slower rates, a Depth of 10.0 gives the richest sound. At faster Rates, high Depth settings sound more like a mild ring modulator effect. Setting the Depth to 1.0 or below creates a "shimmering" sound I haven't heard from any other chorus effects.
- **Chorus (thinner sound).** Use the same settings as above, except change Dual Type to Single. Lower Depth settings are most appropriate at faster rates, and higher Depth settings at slower rates.
- **Pulsing tremolo.** Choose the Dual Type, Mix to 100%, and set Depth to 0.0. Faster Rate settings give a pulsing tremolo effect. However, this reduces the low-frequency response, so it's not as suitable for bass. You can bring the low end back in by changing Mix to a lower percentage, but then the pulsing effect isn't as dramatic.

Tip: Reduce the Rate, and you'll hear clearly how the two chorus paths interact with each other.

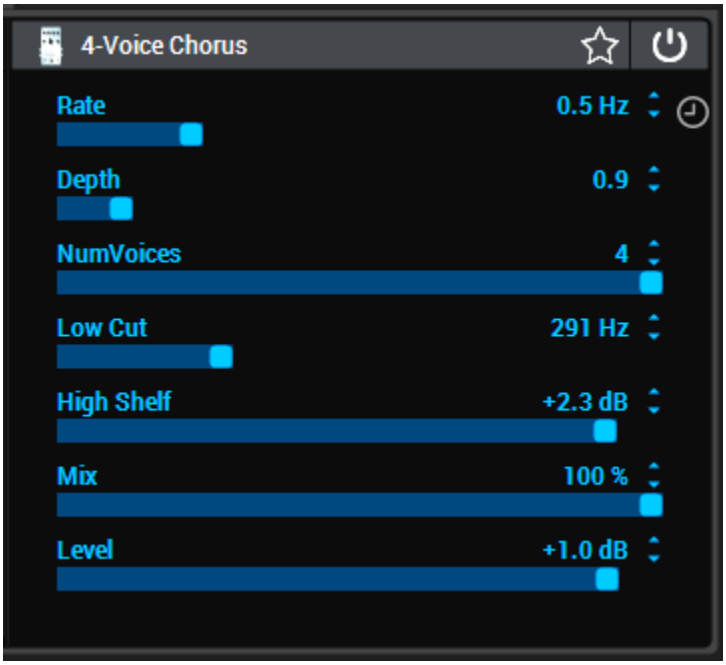
- **Vibrato.** This is a convincing vibrato effect, with just a tiny hint of chorusing. It's turning into my favorite way to produce vibrato with Helix. Select the Single Type, set Mix to 100%, Depth between 1.0 and 2.0, and Rate for the desired vibrato speed.

The interaction among the Type, Depth, Mix and Rate controls produces such different effects that it's worth saving some of these as Favorites.

Tip: I like following the Acoustic Simulator with this chorus.

4-Voice Chorus

This Line 6 original was designed to make people forget about choruses being a stereotyped sound of the 80s. Okay, not really. But it might end up doing that anyway. Few choruses provide four individually modulated voices. More voices give a more diffused, and less common, chorus effect.

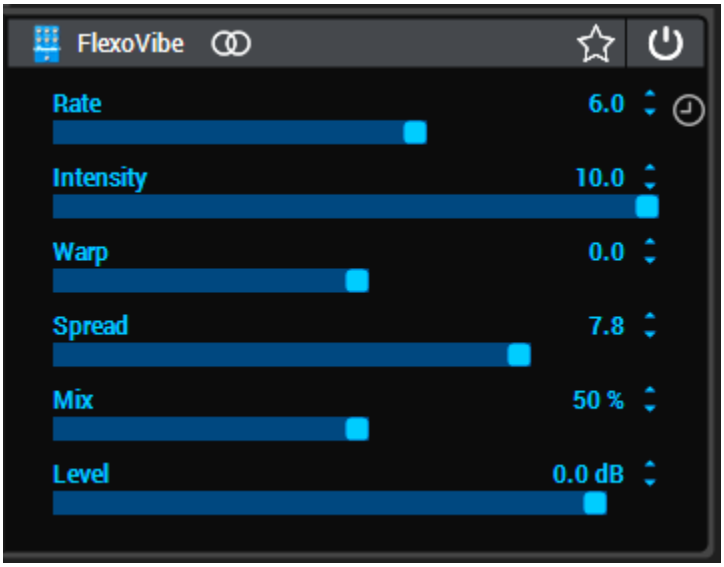


- Rate sets the chorus modulation rate, from 0.1 Hz to 2.0 Hz.
- Depth determines how much the modulation affects the chorus delay time. More depth increases the apparent chorus effect.
- Voices selects 2, 3, or 4 copies of the input. The effect applies modulation individually to each voice. More voices create more pronounced chorusing effects.
- Low Cut reduces the response to low frequencies. The filter curve provides a gentle slope downward from the selected frequency. Sometimes chorus can “blur” the lower frequencies. Because this control affects the chorused sound, not the dry, you can retain the chorus’s brighter tone while reducing the chorus’s low end to retain focus.
- High Shelf provides a mild high-frequency boost or cut, starting at around 2 kHz. Like the Low Cut, this affects only the chorused sound.
- Mix controls the wet/dry proportion, from 0% (no chorus) to 100% (chorus only, no dry sound). Traditionally, chorus units give the maximum effect at 50%. However, with this chorus, higher percentages can give useful effect. The 4-Voice Chorus screenshot above is a good example of a sound that works well with 100% chorusing.
- Level sets the block’s overall output level.

Tip: If you use Helix Native, try this processor on strings, organs, and other signal sources that benefit from more motion and animation.

FlexoVibe

This Line 6 original effect modulates amplitude, phase, and (in stereo mode), panning. If there's any other effect out there like this, I certainly haven't heard it. Line 6 considers it a chorus, but the sound arguably has more in common with a phase shifter. Note that choosing mono or stereo gives very different characters. It's at its best in stereo.



- **Rate** isn't calibrated in Hz. I estimate the slow speed is around 0.1 Hz, and the fastest speed, 4.5 Hz.
- **Intensity** determines the amount of modulation.
- **Warp** controls the LFO waveform, but the waveform also varies according to the Intensity setting. Fig. 2.45 shows the audio output at different settings. The test frequency was 400 Hz. At lower frequencies, the peaks have a downward dip.

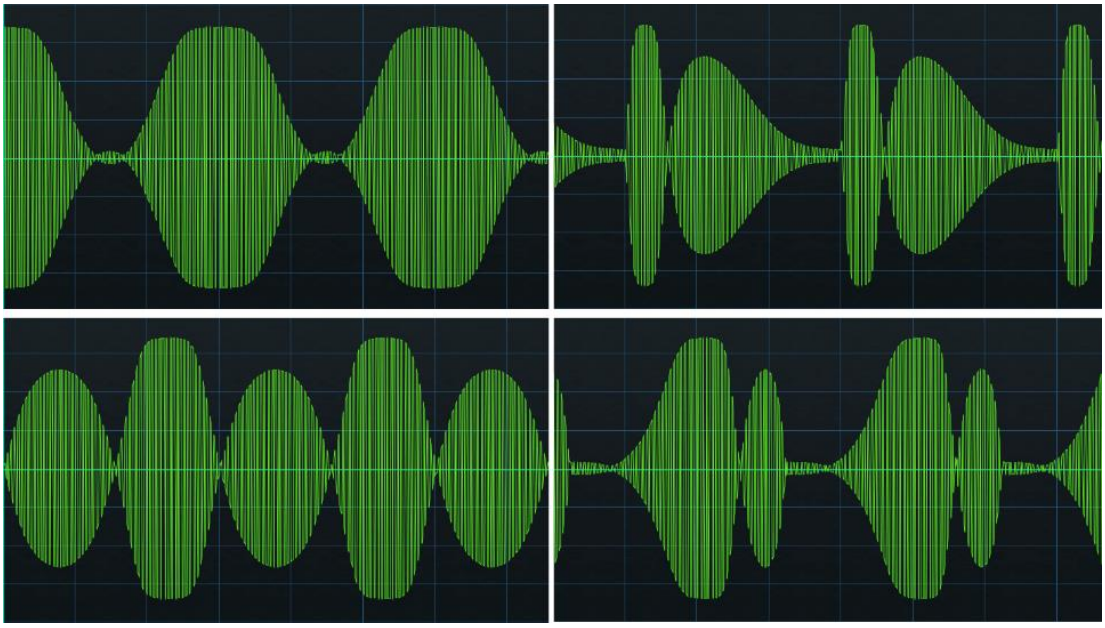


Figure 2.45 Clockwise from upper left: Intensity = 5.0, Warp = 0.0. Intensity = 10, Warp = +1.0. Intensity = 10, Warp = -1.0. Intensity = 10, Warp = 0.0

- **Spread** is available only in stereo mode. It sets the phase offset between the two LFOs. This also affects the stereo image. At 0.0, the image is essentially mono. 10 gives the most pronounced stereo separation. 5.0 splits the difference between those two extremes.
- **Mix** works as expected. 0% has no effect in the mix, while 100% has no dry sound mixed in. The effect is most pronounced at 50%.

The waveforms in fig. 2.45 are more like snapshots that give a general idea of how the modulation affects the audio output. Because phase shifting is involved, the waveform looks different at different frequencies. The main takeaway is how Warp alters not only the shape, but different attack and decay characteristics.

Tip: In either the stereo or mono mode, you can obtain a vibrato effect. Set Warp to 0.0, Spread to 0.0, and Mix to 100%

Delay Blocks

The Delay Blocks include several Line 6 Originals that take echo and delay in new directions. We'll cover common delay parameters first. The sections on the various effects focus on the differences.

- **Time.** With mono delays, this sets the time between delays, in milliseconds or seconds. Time can also sync to note values. With stereo delays, Time sets the delay for the left channel, while Scale (see next) sets the time for the right channel.
- **Scale** sets a delay offset for the right channel, as a *percentage* of the left channel's delay time. For example, suppose the Time is 250 ms and scale is 60%. The left channel's first echo will be 250 ms after a note, and the right channel's first echo will be 250 x 0.6 = 150 ms. From there on, each echo will be 250 ms later (e.g., the left channel's second echo will be 500 ms, the third echo 750 ms, etc. For the right channel, the second echo will be 400 ms, and the third echo 650 ms; see fig. 2.46). Specifying the right channel's delay time as a percentage maintains the time relationship between the delays when synching to tempo, or changing Time. Scale percentages can also give polyrhythmic effects.

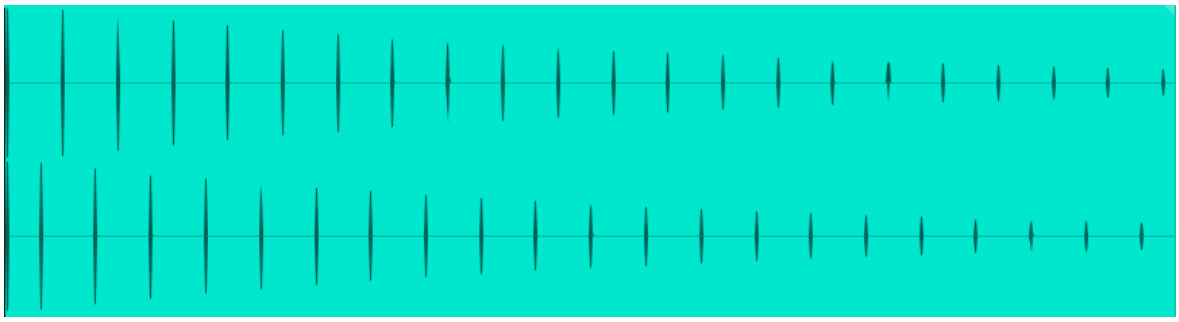


Figure 2.46 The top shows the left channel echo, the bottom shows the right channel with Scale = 60%. In the right channel, after the first echo happens at 60% of the left channel's delay, subsequent right channel echoes are offset by the left channel's Time.

- **Feedback** sends some of the audio output back to the input, so the returning echo creates another echo. More feedback leads to more echoes. With some delays, 100% feedback gives "infinite" echoes.
- **Level** sets the block's output level, which allows for gain-staging.
- **Mix Level** chooses the balance of the dry, unprocessed audio with the delayed audio.
- **Low Cut** reduces low frequency response starting at a specific frequency, from 20 to 500 Hz (below 20 is off), while **High Cut** reduces high frequency response starting at a specific frequency, from 500 Hz to 20 kHz (above 20 kHz is off). They can emulate the bandwidth limitations of older delays.
- **Modulation Mode.** Several delays offer the option to modulate the echoes. The usual choices are Off, Chorus (modulates the delay time with a waveform and mixes in dry signal), and vibrato (modulates the delay time with a waveform, to vary pitch). Vibrato does not mix in the dry signal.
- **Shape** chooses the modulation waveform, which determines how the waveform changes over time. For more information, see Modulation Blocks.
- **Duty Cycle** further alters the wave shape, as also described in Modulation Blocks.
- **Speed** specifies the modulation rate in Hz.
- **Depth** determines how much the modulation affects the signal being modulated.
- **Headroom** is a unique Helix parameter that comes into play with hotter-than-normal input signals. For more information, please see Modulation Blocks.
- **Spread** increases the stereo image width with stereo delays. With some delays, adjusting the stereo spread requires having modulation in Chorus or Vibrato mode.
- With **Trails** on, after bypassing a delay effect or choosing a different Snapshot, any remaining decay from the delay continues. When off, bypassing an effect cuts off any further delays.

Simple Delay, Mod/Chorus Echo, Dual Delay, Ping Pong

The preceding section, Delay Blocks, describes all the parameters for the stereo delays in fig. 2.47. Dual Delay and Ping Pong are stereo-only. Mono versions of Simple Delay and Mod/Chorus Echo don't have a Scale parameter because there's only one delay. Here's what else differentiates them:

- Simple Delay and Dual Delay have a flat delay path frequency response, unlike "character" delays.
- Mod/Chorus Echo adds Low and High Cut filters, as well as chorus or vibrato delay modulation.
- Dual Delay has independent (not scaled) left and right channel delay times. This is great for polyrhythms and creating a stereo image from a mono signal.
- Ping Pong bounces echoes between the left and right channels. Spread determines the panning width.

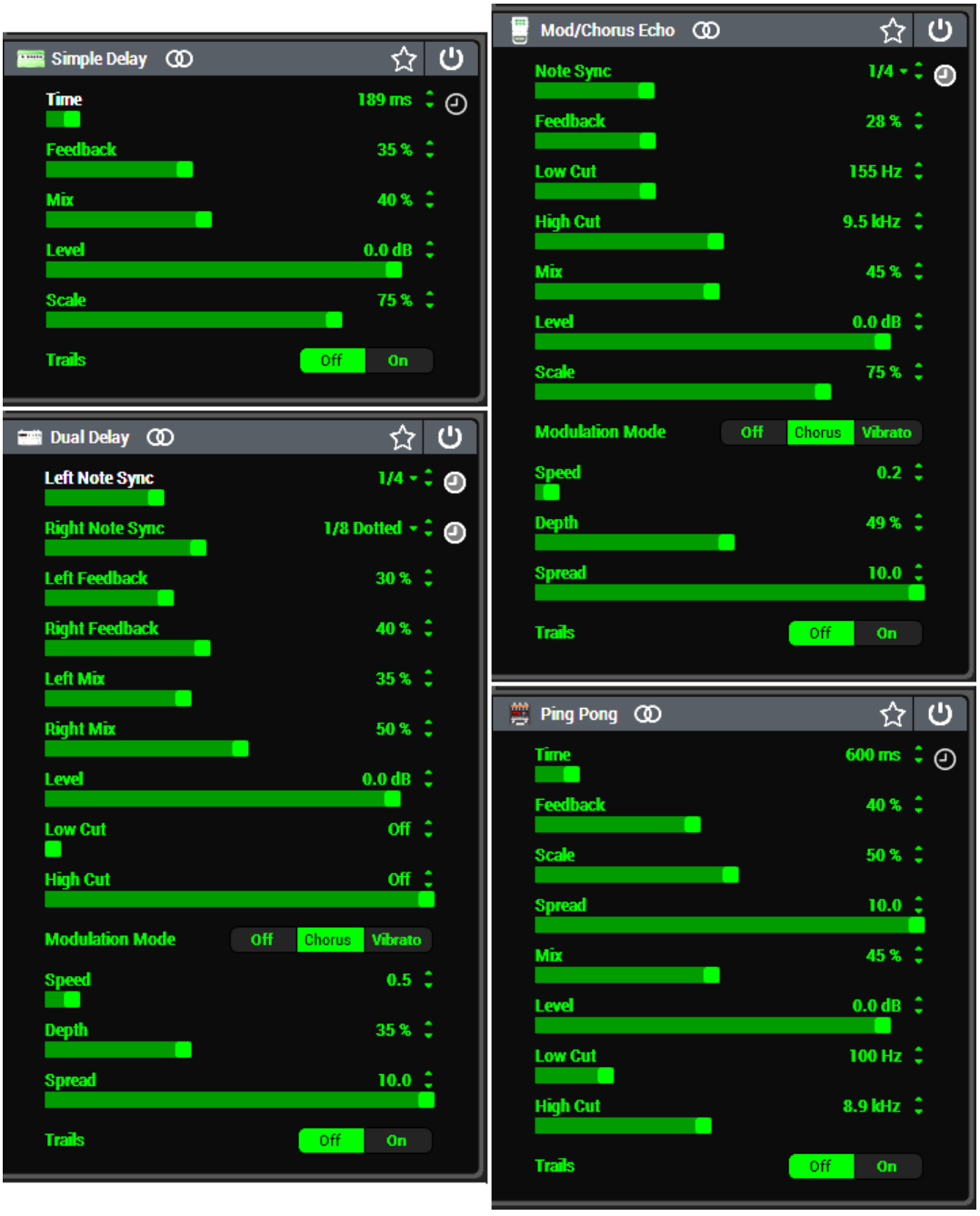


Figure 2.47 These are Helix's four most basic delays.

Multitap 4 and Multitap 6

Fig. 2.48 shows the Multitap 4 delay's parameters (the interface is split in two halves to fit better on the page). The MultiTap 6 replaces the MultiTap 4's modulation section with two more taps and omits the Diffusion control. Some DAWs call tapped delays "groove delays," because they enable rhythmic possibilities.

Figure 2.48 4-Tap Delay parameters.

Scaling is an important part of the Multitap delays. Fig. 2.49 shows a practical example.

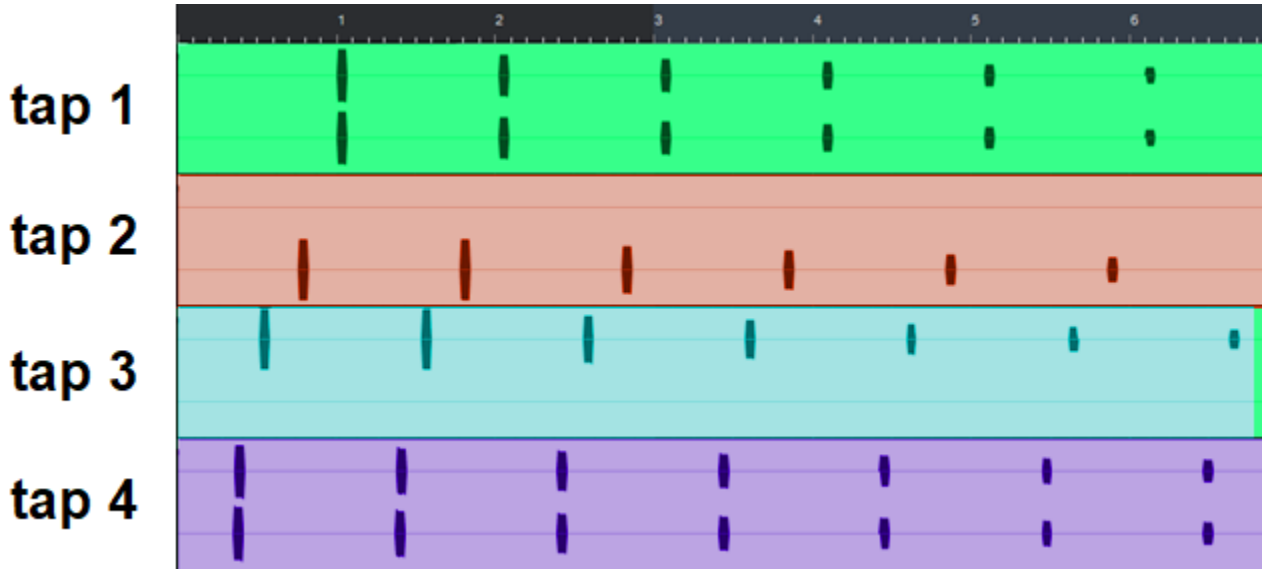


Figure 2.49 Scaling applied to four taps.

The **Time** parameter sets the master time. The **Scale** parameter scales the tap times in relation to the master Time parameter, and **Pan** can place the taps anywhere in the stereo field. The scaling protocol is the same as described in Delay Blocks. Note that the Multitap Delays are not "character" delays, so the delay path has a flat frequency response.

In fig. 2.49, the Time is set to 1.000 seconds:

- The top track shows repeats from Tap 1, set for 100% Scale and panned to center. (Interestingly, note how the timing drifts slightly later over time.)
- The second track down shows Tap 2 set for 75% Scale and panned right.
- The third track down shows Tap 3 set for 50% Scale and panned left.
- The fourth track down shows Tap 4 set for 33% Scale and panned to center.

Multitap 4 delay is the only delay with a Diffusion control, which operates like the diffusion control in reverbs. When set to maximum, the echoes are more "spread out" and sound less distinct—more like reverb than echo. With minimum diffusion, you can hear the individual echoes more clearly (fig. 2.50).

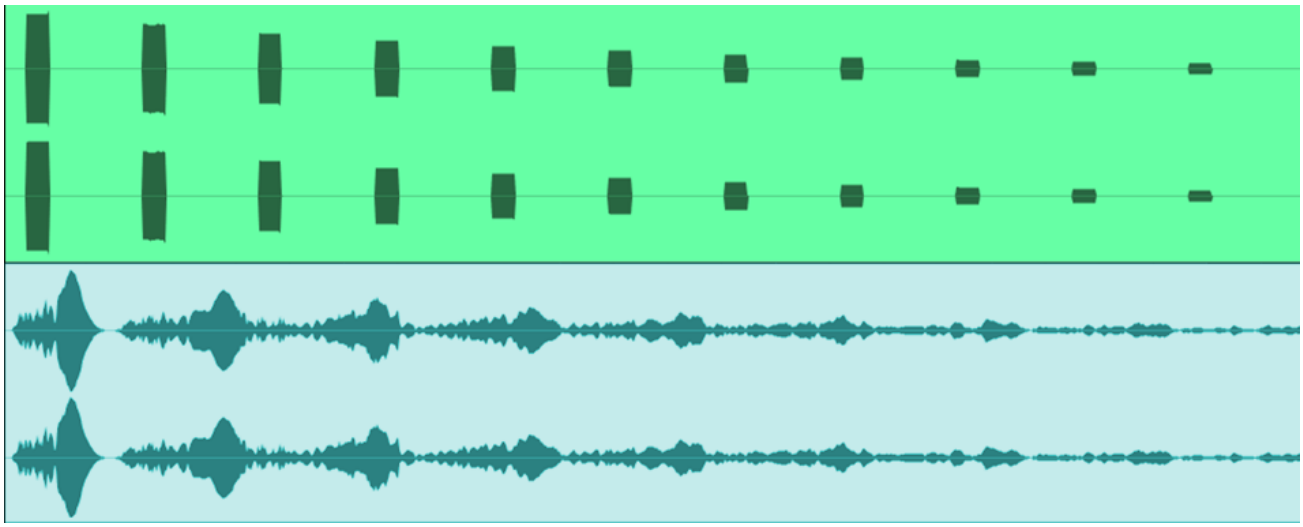


Figure 2.50 The top track shows echo applied to a short, sharp pulse. The bottom track shows the same sound with maximum diffusion. This gives the sound a more reverb-like quality.

Tip: High diffusion settings are used mostly with percussive sounds to prevent the "marbles bouncing on a steel plate" effect that happens with lots of discrete reflections. Low diffusion settings are common with sustained sounds, so that the reverb is a little thinner, and doesn't obscure the sound being processed. High diffusion settings are also excellent when using the Multitap 4 to create room sounds with very short delays (e.g., under 40 ms or so).

Sweep Echo

Consider this as two effects folded into one user interface: A standard delay, and an LFO-swept, multi-mode filter (Lowpass, Bandpass, Highpass) that processes the delayed audio (fig. 2.51).

Figure 2.51 Sweep Echo is two effects in one—delay (left), and LFO filter (right).

The echo part is straightforward. Most of the controls are covered in the section Delay Blocks, except for:

- **Scale** in the filter section probably belongs in the echo section, because it does what Scale does for other delays—sets the right delay time as a percentage of the left (main) delay time.
- If Scale is set to 100%, **Spread** pans the delays to center at 0.0, and hard left and right at 10.0. With lower Scale values, the stereo separation remains intact, but higher Spread values differentiate the filter settings more for the left and right channels.

The LFO-based filter section has a lot in common with wah pedals:

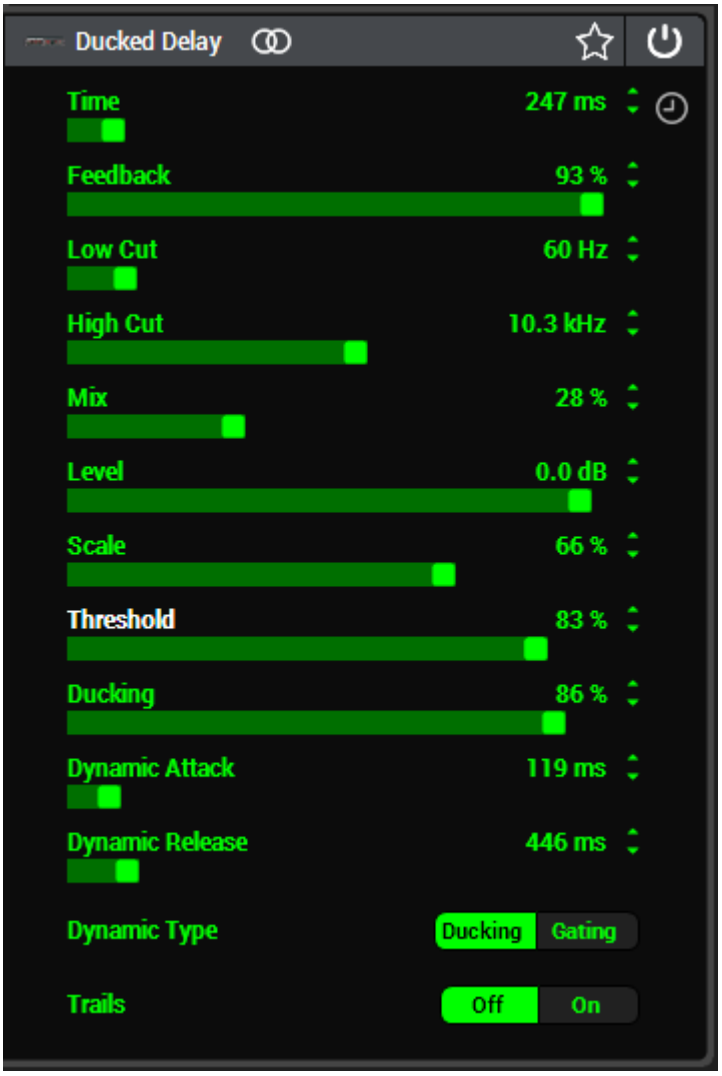
- **Start Frequency** sets the lowest filter frequency, like the toe back on a wah pedal.
- **Range** sets how high the filter frequency can go.
- **Resonance** raises or lowers a resonant peak around the filter's cutoff frequency (or center frequency in Band Pass mode).
- The other parameters are covered in the Modulation Blocks section, but **Rate** behaves differently. Although it sets the LFO speed, 0.0 "freezes" the current filter frequency (set by the Start Frequency and Range parameters). Changing Rate to 0.0 also freezes the filter frequency at the time of the change, although there's some lag time between making the change, and having it take effect.

Tip: Defeating the echo creates a bonus effect—an LFO-controlled filter. Set the delay time to 0.0 ms, with 0% Feedback and Mix to 100%.

Ducked Delay

This powerful effect combines delay, gating, and ducking. To get the most out of this versatile effect, it's necessary to understand how several of the parameters interact. Its two types of operation are:

- **Ducking** reacts to the level of your playing. It mutes the delayed audio while you're playing, then releases muting when you stop playing. So, if the delay time is long and there's feedback, the echoes appear after you stop playing. Then, they trail off.
- **Gating** does the opposite—it mutes the delayed audio when you *aren't* playing. You hear any delay effect only while you're playing.



Let's look at the parameters in depth.

- **Threshold.** This can be confusing, because raising a traditional gate threshold control makes the gate *less* sensitive to input signal levels. With the Ducked Delay, Threshold acts more like a sensitivity control. The higher the Threshold percentage, the more likely that ducking or gating will occur. For example, with a low-level signal and low threshold, gating will not happen. As you raise the Threshold, lower-level signals can duck or gate the delayed audio.

- **Ducking.** When ducking occurs, higher parameter percentages attenuate the delayed audio more. With Gating, when the gate is closed, lower percentages attenuate the delay more.
- **Dynamic Attack.** Increasing the attack time fades in the ducking or gating effect. For example, to duck or gate immediately when playing starts, set the shortest time.
- **Dynamic Release.** With ducking, increasing the release time fades in the delay's audio after you stop playing. With gating, increasing the release time fades out the delay's audio after you stop playing. For example, when ducking, if you want the delay to start immediately after you stop playing, choose the shortest time.

Here are some Ducked Delay applications:

- When using lots of feedback, ducked delay can keep the ongoing delays from competing with your playing. But in the spaces between phrases, the feedback can bloom to its full strength.
- Gating isn't as common, but one application is tightening tempo-synched delay. While you play, the tempo-synched delay plays along with you but when you stop, the delay stops as well.
- For ambient-type music, you can have long delay times with lots of feedback, and duck slightly as you play so the delay doesn't go away completely. When you stop playing, the cascade of sound resumes.

Reverse Delay

The parameters are identical to the Mod/Chorus Echo, except there's no Scale control because there's only one delay. Spread matters only if **Modulation Mode** is set to Chorus or Vibrato.

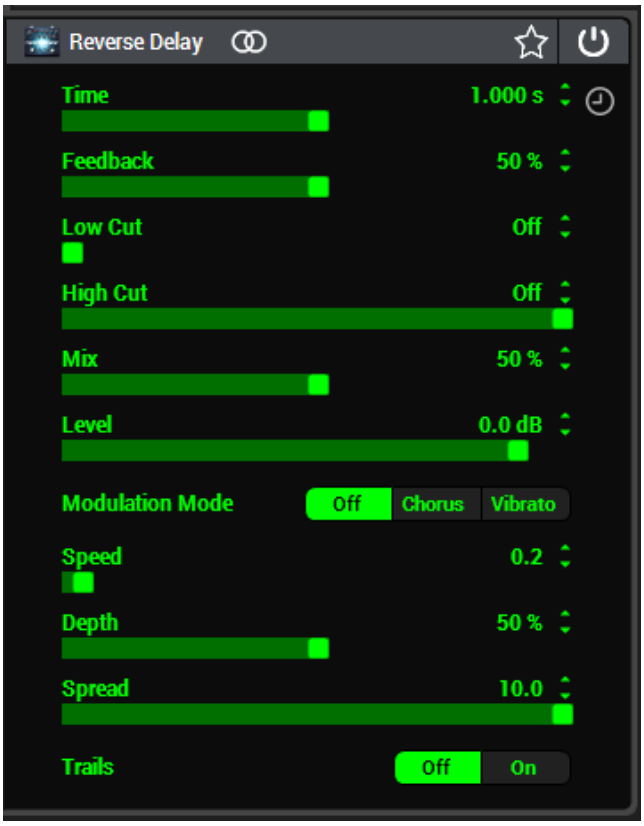


Fig. 2.52 shows the difference compared to standard delay. The original note is purple. The reversed decays are black. It's a novel effect.

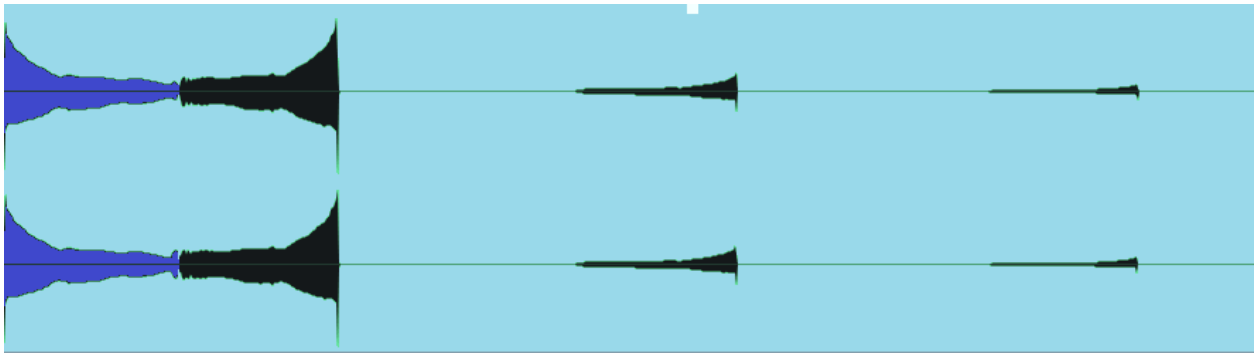
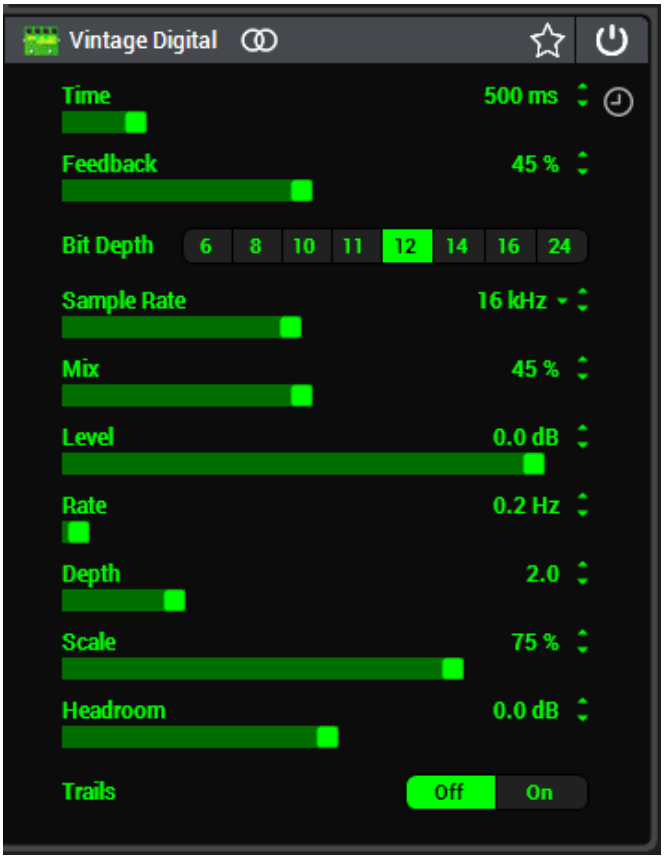


Figure 2.52 With reverse delay, the repeats play "backwards."

Vintage Digital

Vintage Digital is a bit of a misnomer. It not only emulates the gritty sound of older digital delays and can venture into lo-fi territory, but it can also produce the high-fidelity, full-bandwidth delay of today's effects.



The parameters are like other delays. Here are the exceptions:

- **Bit Depth** sets the audio resolution for echoes. Higher bit depths reproduce the delay's echoes more accurately. For lo-fi effects, the 6-bit setting adds background digital artifacts.
- **Rate** and **Depth** alter the delay time periodically, to create vibrato-like variations.
- **Sample Rate** sets a ceiling on the high frequency response. In digital's early days, analog-to-digital and digital-to-analog converters had lower bit resolutions than today's 24-bit converters. Memory was expensive too, so these delays often used lower sample rates to avoid having to store too many samples. Lower sample rates reduced the high-frequency response (fig. 2.53).

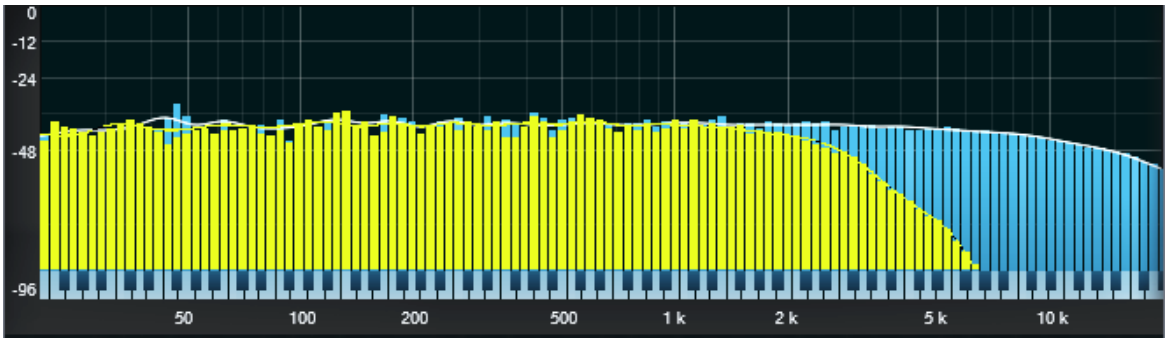
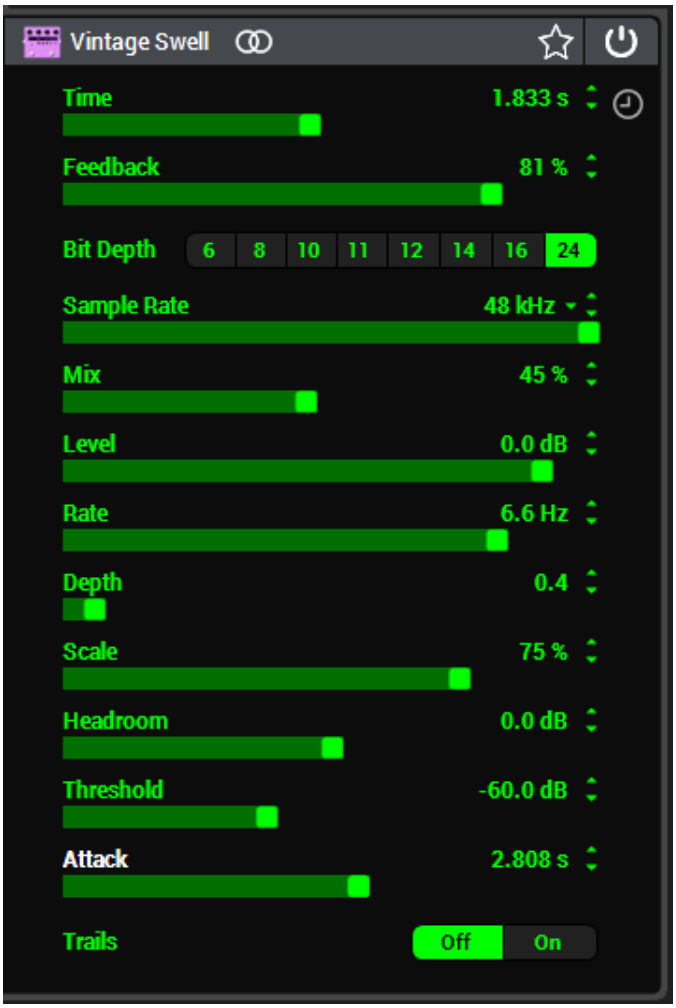


Figure 2.53 Frequency response with 8 kHz sample rate (yellow), and 44.1 kHz (blue).

Vintage Swell

This effect is like combining Autoswell and Vintage Digital, so you don't have to use up two blocks to create one effect. Like the Autoswell, after a brief period of silence, hitting a note or chord whose level exceeds the Threshold causes the audio to ramp up in level. The Attack parameter sets the ramp time. The delay is not of the original input, but the "swelled" audio.



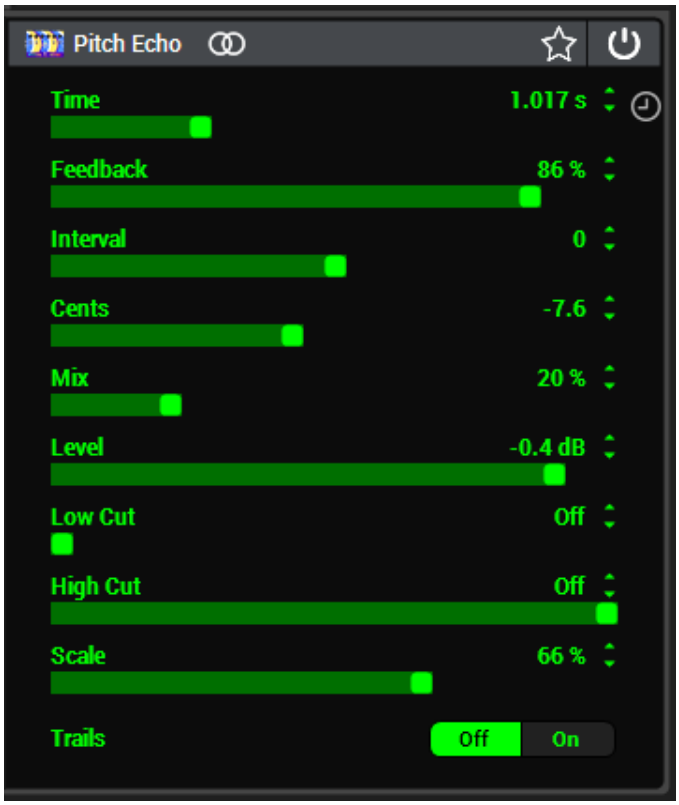
The parameters are identical to the Vintage Digital effect, with two exceptions:

- When the guitar's level drops below the **Threshold**, the Autoswell action resets, and the level is ready to ramp up again. To hear the next note you play, its level needs to exceed the Threshold. If the effect seems silent, lower the Threshold level to a negative enough value for the audio level to exceed it.
- **Attack** is the time it takes to ramp up from silence to full volume.

Try this effect with sustained, distorted chords, a long Time parameter, plenty of Feedback, and a long Attack time. The chord will fade in, and the echoes behind it will fade in as well, creating an evolving, rich texture.

Pitch Echo

Echoes are usually at the same pitch as the delayed signal, but the Pitch Echo transposes the delayed sound. Aside from **Interval**, which sets the transposition amount, the parameters are like a conventional echo.



The following applications make use of the interval parameter.

- One common use, sometimes called a “bell tree” effect, is setting an Interval, increasing Feedback, and hitting a single note or quick upstroke/downstroke. The Interval amount transposes each successive echo. For example, with a +1 semitone Interval, the first echo will be 1 semitone higher, the second echo 2 semitones higher, the third echo semitones 3 semitones higher, and so on, until the feedback dies out (or the pitch goes higher or lower than what Helix can handle).
- For a rockabilly tape echo sound, set the **Delay** time to around 135 ms, **Feedback** to 0%, **Cents** to around +7 or +8, and **Mix** to 30%. The slight pitch transposition emulates the less-than-perfect tape transport speeds of the 1950s.
- For subtle flanging that sounds less like an “effect” and more like “animation,” set **Feedback** to 0, **Mix** to 50%, **Time** to around 5 ms, and **Cents** to about -6.0.

Transistor Tape

This is a model of the solid-state successor to a popular tube-based, tape echo unit. Initially, I was concerned there weren't Low and High Cut filters to emulate the bandwidth reduction that happens with high feedback amounts (caused by tape and tape head issues). Fortunately, the emulation builds that sound into the model itself (fig. 2.54)—and even nails what happens when you turn up feedback so high it self-oscillates, which turns the sound into a sea of noise.

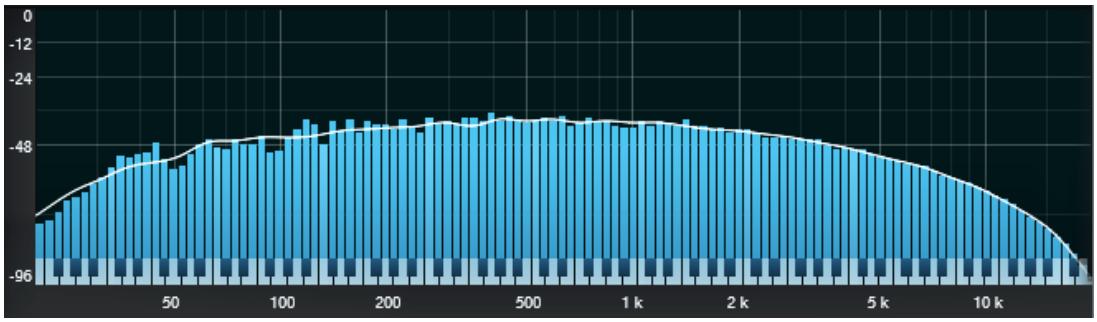
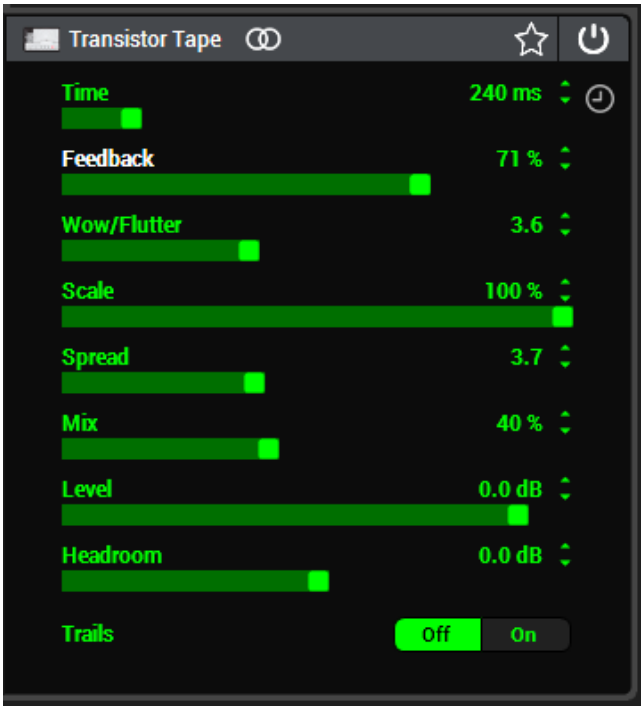


Figure 2.54 The Transistor Tape frequency response is representative of tape-based echo units, with rolloffs at the high and low frequencies.

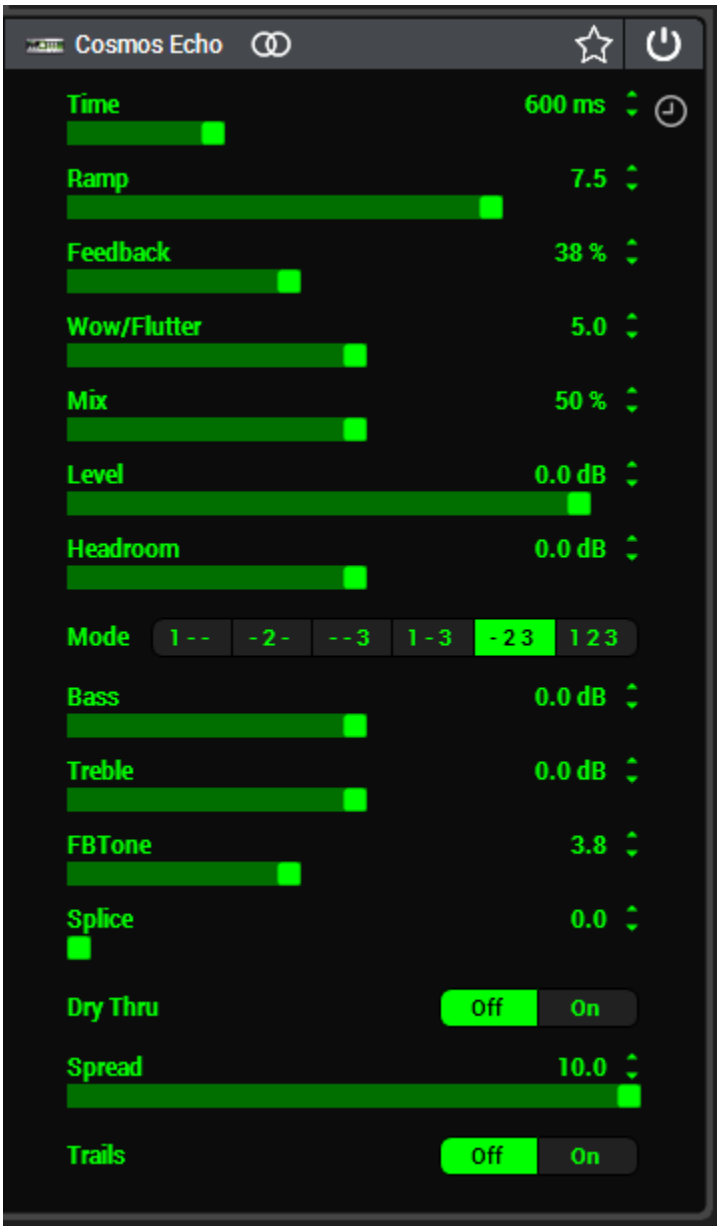
The effect's basic parameters are like those in other effects, except for the **Wow/Flutter** parameter. Far from being a gimmick, it adds some "meat" to the sound that's *not* like chorusing-style modulation. I didn't expect to like it—I never liked wow and flutter with multitrack tape recorders—but it works well here and adds an element of authenticity.



You owe it to yourself to turn up feedback all the way at some point (with the monitors initially turned down!). The glorious sound that peaks around 700 Hz launched a zillion alien sound effects in 1950s science-fiction movies. Or maybe you'll feel what it's like to lie down in a field populated by a horde of angry, and perhaps carnivorous, crickets. In either case, it's kind of impressive that Line 6 duplicated this sound.

Cosmos Echo

This echo effect emulates a vintage tape-based device with three playback heads, which played back sounds from a tape loop. The loop played continuously and was driven by a capstan motor. The original unit on which this is based was known for unpredictable behavior that many found (and still find) endearing. It's by far the quirkiest of the Helix delays.



Although many of the parameters may *seem* familiar, they have some interesting twists:

- **Time** is the master time parameter. It sets the delay when playing back through Head 3. Head 2's delay is 2/3 the Time parameter, and Head 1's delay is 1/3 the Time parameter. See Mode for more information.
- **Ramp.** With the original unit, changing Time wasn't instantaneous because the change in tape speed was subject to a mechanical lag. Ramp sets the rate of change when you alter the Time parameter. 0.0 gives the fastest rate of change, 10.0 the slowest.

- **Feedback** is a highly variable parameter that's not like the echoes of analog or digital delays. The tone, bandwidth, and sound quality vary considerably as the feedback keeps going. It's also possible to drive the Cosmos Echo into self-oscillation where repeats go on forever, even at lower values (e.g., 50%). You needn't always go up to 100% to enter the world of feedback insanity.
- **Wow/Flutter** adds pitch variations to the virtual tape, but if there's feedback and some of the echoes "land" on previous echoes, the wow and flutter add a chorusing-type effect. This doesn't happen with the value at 0.0.
- **Mix** sets the ratio of dry and processed sound. See Dry Thru for more details.
- **Level** acts like a standard output level control. It may be the only parameter in this effect that's relatively normal.
- **Headroom** works the same way as described in Modulation Blocks.
- **Mode** chooses which head(s) will play back echoes. For example, suppose the Time is set to 600 ms. With Mode 1 - 3, you'll hear a 200 ms echo from Head 1 and a 600 ms echo from Head 3. With Mode - 2 - 3, you'll hear a 400 ms echo from Head 2 and a 600 ms echo from Head 3. Each Mode creates a different type of repeating pattern. Adding feedback to the more complex modes creates some novel effects.
- **Bass** lifts or attenuates the low frequencies. **Treble** does the same for the high frequencies (fig. 2.55).

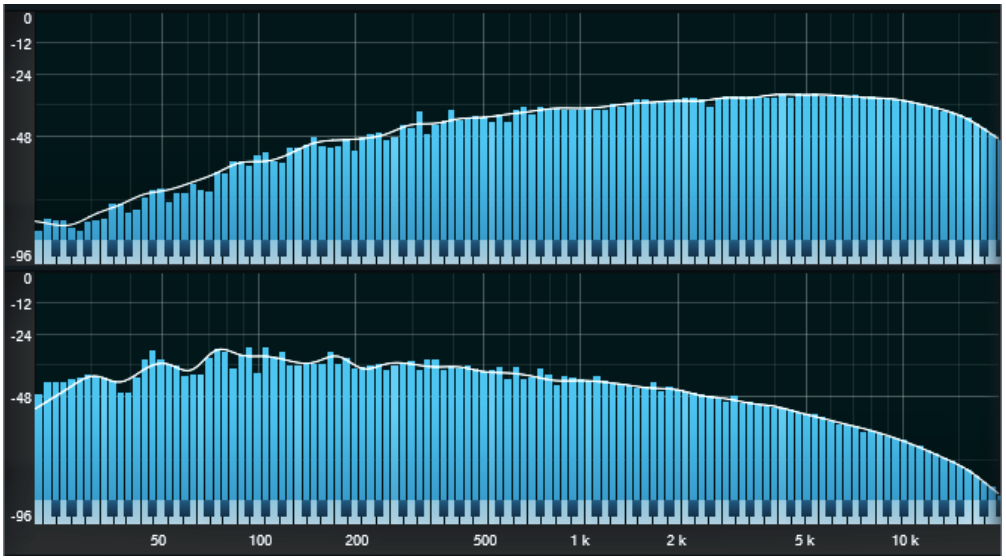


Figure 2.55 Top: Bass -18 dB, Treble +18 dB. Bottom: Treble -18 dB, Bass +18 dB.

- **FB Tone** alters the bandwidth as feedback continues (fig. 2.56). When feedback first starts, the bandwidth is at its widest. As the feedback continues, the bandwidth narrows. At lower FB Tone settings, the bandwidth narrows, with a center frequency around 400 Hz. At higher FB Tone settings, the bandwidth extends into a higher frequency range, with a somewhat higher level. However, bear in mind this effect changes over time, and depends on the amount of feedback.

  Fig. 2.52 shows what happens on the 20th repeat of a single head, with feedback set to 70%. Bass and Treble are at 0.0 dB (FB Tone is influenced by these controls). Note the difference with FB Tone set to minimum and maximum. Although this effect can vary, the general rule is that high FB Tone values give more bandwidth as the feedback continues.

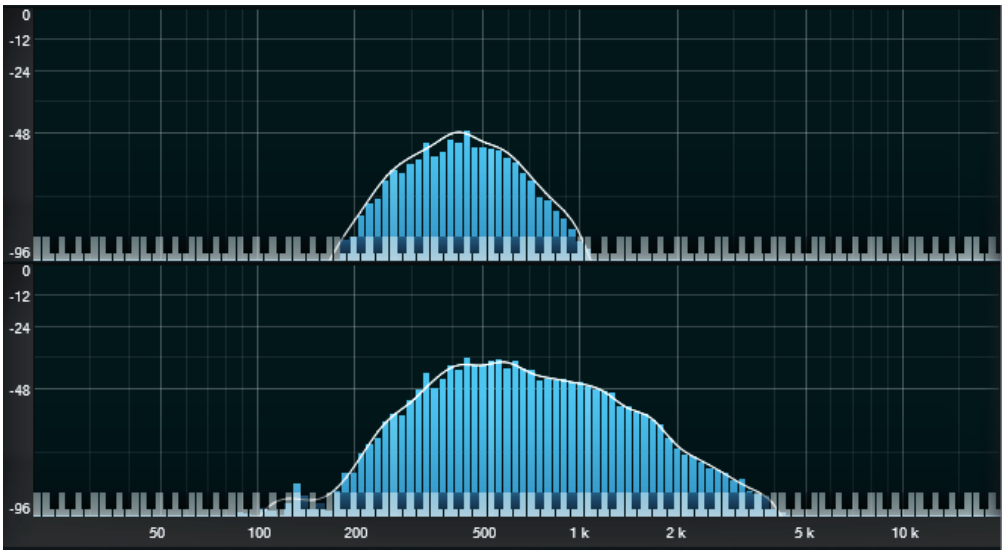


Figure 2.56 FB Tone at 0.0 (top) and at 10.0 (bottom). These spectra were generated by a single head after 20 feedback repeats, from a short pink noise burst.

- **Splice.** To create a tape loop, it was necessary to splice together the piece of tape's beginning and end. The Splice parameter emulates the glitch that occurred when the splice passed over a tape playback head. 0.0 selects no splice. Higher values increase the glitch's severity.
- **Dry Thru.** When Off, all incoming audio goes through the effect's electronics. With Dry Thru On, the Mix parameter varies the ratio of effect to a *paralleled* dry sound. The effect is subtle, and relates mostly to slightly less high-frequency response with Dry Thru off.
- **Spread** does stereo imaging, but it's understated—don't expect echoes ping-ponging back and forth across your head.

Harmony Delay Stereo

The Harmony Delay parameters impart pitches to echoes. This is another one of those effects that has so many parameters, I had to split it so it could fit on the page. This effect works best with individual notes, although nothing prevents using it with chords.

Feedback, **Mix**, **Level**, **Low Cut**, and **High Cut** are all familiar, but that leaves a lot of explaining to do for the other parameters.

The unique concept is that there are three echoes—the original note you play (the root note), and two additional Voices. The Voices can be at different pitches. You need to enter the song's key and choose an appropriate scale to which the Voice pitches can conform. Here's how to tweak the various parameters:

- **Time.** When you hit a note, Time sets the master delay, after which the Root note and Voices repeat. However, you can alter the delay time for the two Voices compared to the root by using the Voice 1 or Voice 2 Scale parameter (see later).
- **Key.** Specifies the song's key.
- **Scale** selects among Major, Minor, Major Pentatonic, Minor Pentatonic, Harmonic Minor, Melodic Minor, Whole Tone, and Whole Diminished (see Tech Talk below for the scale notes and intervals).
- **Voice 1 Shift** chooses the first repeat's scale note for the selected Key. For example, in the key of A Major, +2 would pick the scale's second note (C#). +4 would pick the scale's 4th note, or E.
- **Voice 2 Shift** does the same as Voice Shift 1 for the second repeat.
- **Voice 1 Level** and **Voice 2 Level** set the level for their respective repeats.
- **Voice 1 Pan** and **Voice 2 Pan** set the stereo field position for their respective repeats.
- **Voice 1 Scale** and **Voice 2 Scale** do not relate to the key's scale. Instead, they scale the delay times for the two Voices compared to the master Time setting (like other delays with a Scale control).
- **Root Level** sets the level for the root note repeats.
- **Root Pan** positions the root note repeat in the stereo field.

Tip: If you want the repeats to be only from the two voices, so that you hear the root note without echoes, set Root Level to 0.0. Then, set Mix for the desired blend of the dry root note with the two Voice repeats.

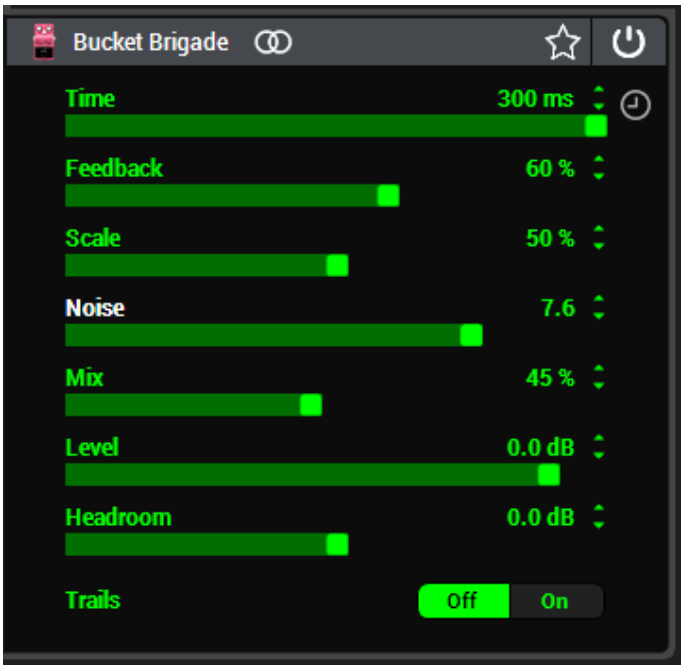
Tech Talk: Scales Used in the Harmony Delay

For a refresher on the notes in the Harmony Delay's scales, the following refers the scale notes to E. The intervals between notes are in parentheses.

Major: E • F# • G# • A • B • C# • D# • E (2-2-1-2-2-2-1)
Minor: E • F# • G • A • B • C • D • E (2-1-2-2-1-2-2)
Major Pentatonic: E • F# • G# • B • C# • E (2-2-3-2-3)
Minor Pentatonic: E • G • A • B • D • E (3-2-2-3-2)
Harmonic Minor: E • F# • G • A • B • C • D# • E (2-1-2-2-1-3-1)
Melodic Minor: E • F# • G • A • B • C# • D# • E (2-1-2-2-2-2-1)
Whole Tone: E • F# • G# • A# • C • D • E (2-2-2-2-2-2)
Whole Diminished: E • F# • G • A • A# • C • C# • D# • E (2-1-2-1-2-1-2-1)

Bucket Brigade

The Bucket Brigade has a characteristically warm sound. The parameters are like other delays, except for the Noise parameter.



BBDs had a sound quality that could become ruder and grainier as the delays got longer, but Helix's modeled noise is more polite than older BBDs. (Maybe Line 6 didn't want people to return the Helix, thinking it was defective because some of the effects were "noisy.") You can hear the noise at higher **Mix** settings for the delayed sound and with more **Feedback**, but it's still fairly subtle. Otherwise, this is your basic analog delay, along with a second delay and its associated **Scale** parameter.

Tech Talk: Why BBDs Sound "Warm"

BBDs are known for their warm repeats, but this was due to a technological limitation, not a design goal. A high-frequency clock governed the speed of moving electrical charges from cell to cell in the BBD, which created delay. Longer delays required a lower-frequency clock, and it became more likely that as a result, an audible, high-frequency clock signal would be audible. The solution was to include a sharp high-cut filter to remove a lot of high frequencies (fig. 2.57), thus keeping the clock "whine" out of the audio output.

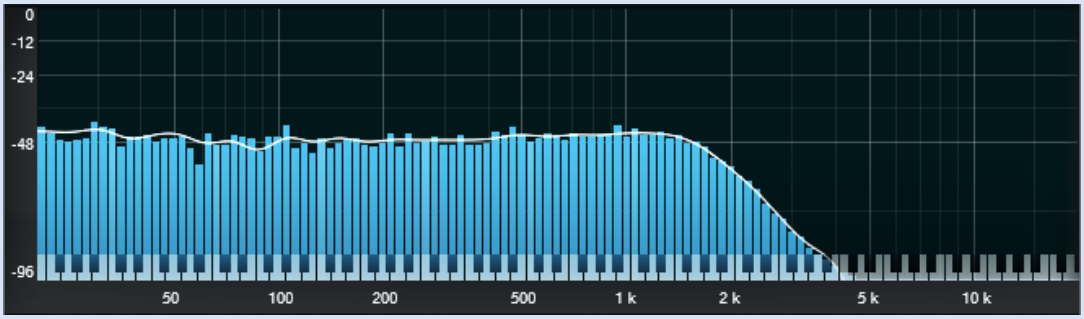
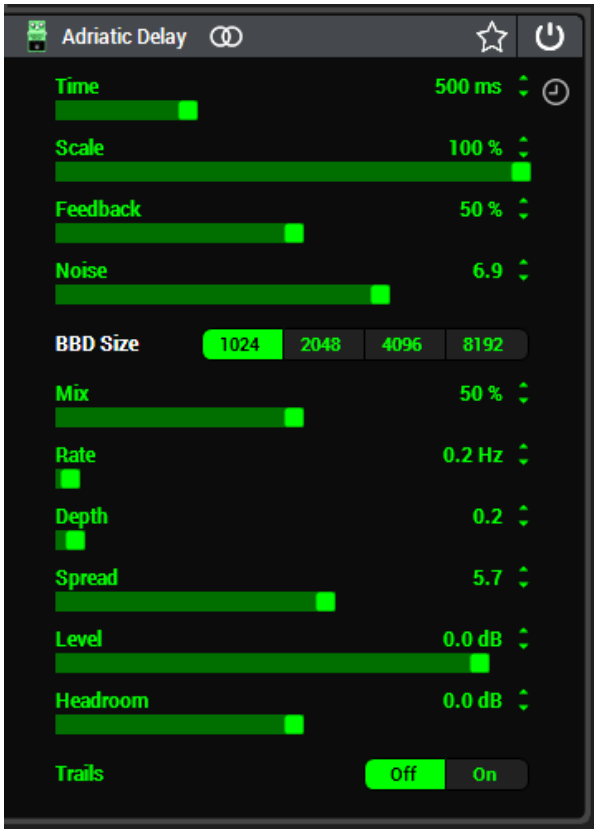


Figure 2.57 Bucket Brigade delay path response with 300 ms delay. The sound is slightly brighter with shorter delays.

Adriatic Delay

This expands on the Bucket Brigade. The main differences are greater options for tailoring the stereo imaging, more delay possibilities, the potential for grittier sound, and modulation. It's as if a modding fanatic had applied a virtual soldering iron to the Bucket Brigade delay. Because the Bucket Brigade is like a subset of the Adriatic Delay, the Bucket Brigade may seem redundant. However, it takes more effort to edit the Adriatic Delay's more evolved set of parameters.



The following parameters aren't in the Bucket Brigade:

- **Rate** and **Depth** add delay time variations that create vibrato-like variations.
- Higher **Spread** values create a wider stereo image.
- **BBD Size.** More cells in a BBD-based device allowed using higher clock frequencies for a given amount of delay (see the previous Tech Talk). Not needing to filter highs so severely results in a better frequency response (fig. 2.58). 1028 gives the least high-frequency response. 8192 gives the most.

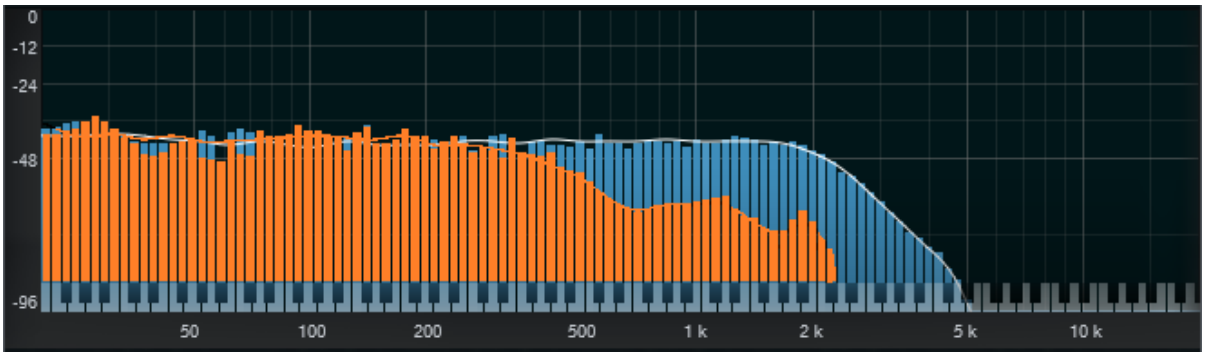
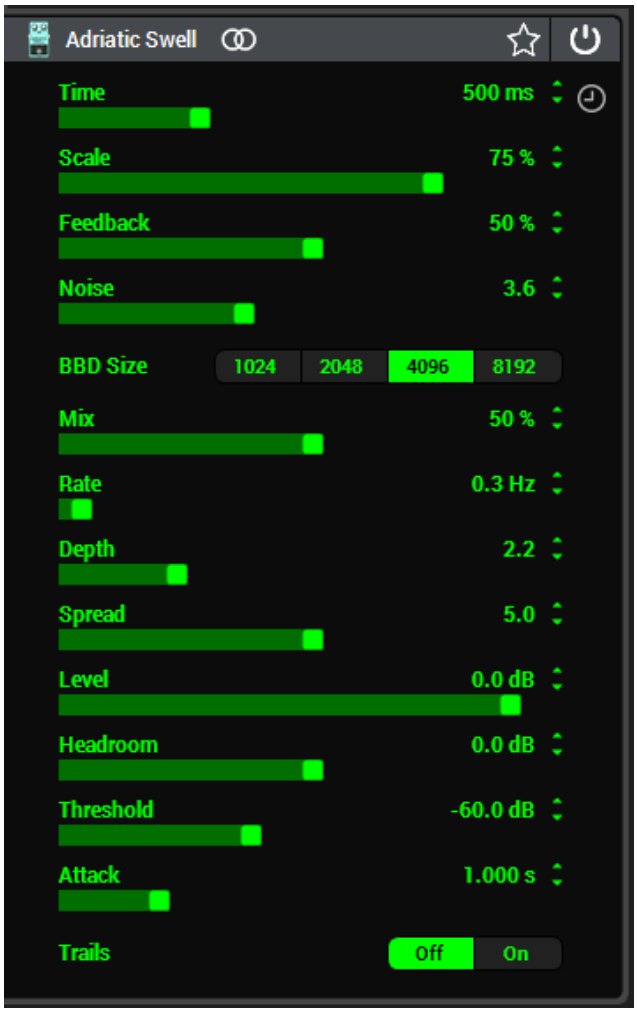


Figure 2.58 Frequency response with BBD Size parameters of 1028 (orange) and 8192 (blue).

Adriatic Swell

This effect is conceptually like Vintage Swell, because it combines Autoswell and Adriatic Delay in a single block. To recap, if you hit a note or chord after a brief period of silence, and the level exceeds the Threshold, the audio ramps up in level. The Attack parameter sets the time for the signal to ramp up. The delay is not of the original input, but the "swelled" audio. This kind of processing excels for evocative, lush, sustained sounds.



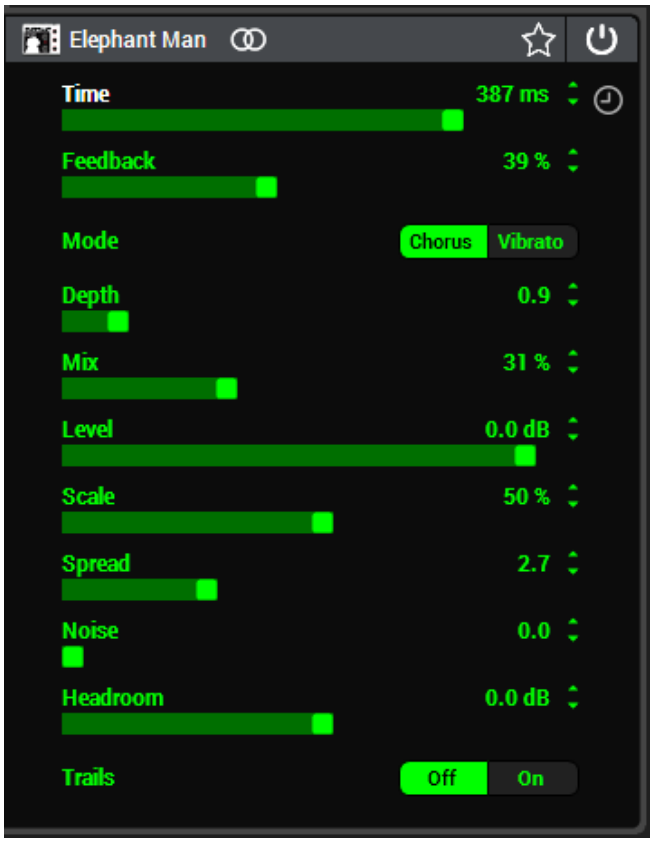
The parameters are identical to the Adriatic Delay effect, with two exceptions:

- When the guitar's level drops below the **Threshold**, the Autoswell action resets, and the level is ready to ramp up again. To hear the next note you play, its level needs to exceed the Threshold. If the effect seems silent, lower the Threshold level to a negative enough value for the audio level to exceed it.
- **Attack** is the time it takes to ramp up from silence to full volume.

The same comments about the Vintage Swell apply to using this effect. I like playing long, sustaining chords through a distorted amp, long delay times, and lots of feedback.

Elephant Man

This effect is based on not only one of my favorite effects, but one of the few effects I used live that I didn't design. There was a certain liquid, lush quality to it that went beyond the usual delay units of that pre-digital era. As with the Bucket Brigade and Adriatic Delay, there's a **Noise** parameter for a dirtier sound. All the other parameters have been covered before. However, Mode has no "off" position—a 0.0 Depth setting cancels the Chorus or Vibrato effect. There are no Low Cut or High Cut filters.



Part of the warm sound is due to output filtering that keeps the virtual clock noise out of the delayed path output (fig. 2.59). At maximum delay, the high frequency response starts dropping off around 3 kHz. The response improves only slightly with shorter delays.

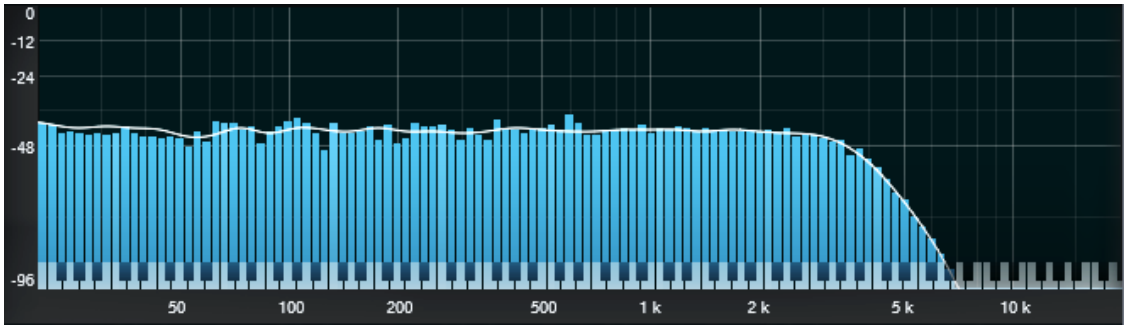
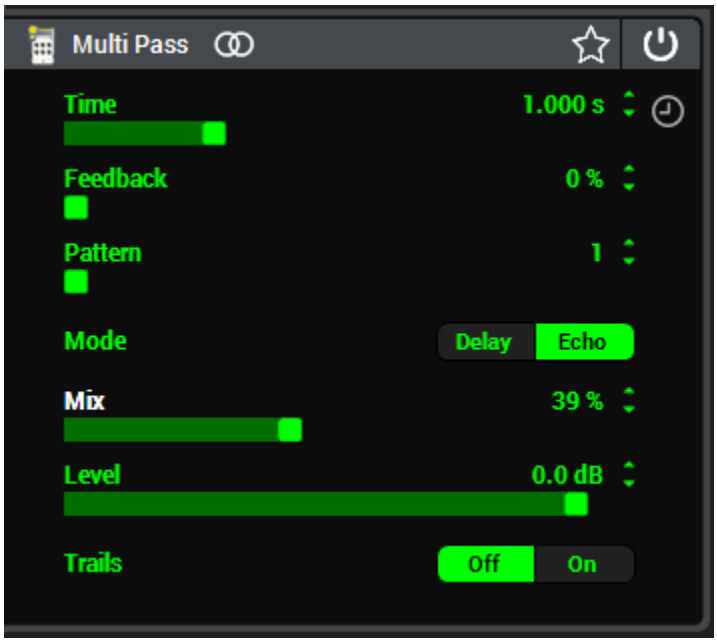


Figure 2.59 Frequency response, delayed sound only.

Multi Pass Delay

This effect's operation may not be intuitively obvious—but we'll take care of that! The combination of delay and filtering can produce a wash of sound, and the rhythmic synchronization adds another flavor. Also note that this is an incredible effect for instruments other than guitar, like drum tracks, hand claps, arpeggiated synthesizers, and much more. This is just one reason why musicians other than guitar players use Helix.



The Multi Pass Delay is a delay with 6 scaled taps. Each tap has a filter. The bass tap appears to be a lowpass filter, the high tap a highpass filter, and the others are relatively low-resonance bandpass filters. What makes this effect so interesting is that the taps are sequenced, in one of several Patterns. Helix doesn't give the taps names, but I've named their frequency ranges to make later explanations easier. The tap frequencies are:

- Bass (lowpass filter)
- Upper bass (250 Hz)
- Low mid (500 Hz)
- Mid (1 kHz)
- High mid (2 kHz)
- High (highpass filter)

When you play a note, the Multi Pass plays each tap in sequence. For example, with a delay time of 1.5 seconds, after the initial note hits, the taps play back at the following times (the time scaling is fixed):

- 250 ms
- 500 ms
- 750 ms
- 1000 ms
- 1250 ms
- 1500 ms

Which tap plays back depends on the Pattern you've chosen. The Time parameter sets the Pattern duration. The Pattern sequences are (see the tap names above):

- 1: Bass > upper bass > low mid > mid > high mid > high
- 2: High > high mid > mid > low mid > upper bass > bass
- 3: Bass > mid > upper bass > high mid > low mid > high
- 4: Bass > low mid > high mid > upper bass > mid > high
- 5: Bass > high > upper bass > high mid > low mid > mid
- 6: Low mid > mid > upper bass > high mid > bass > high
- 7: High > low mid > high mid > upper bass > mid > bass
- 8: High > mid > upper bass > high mid > low mid > bass

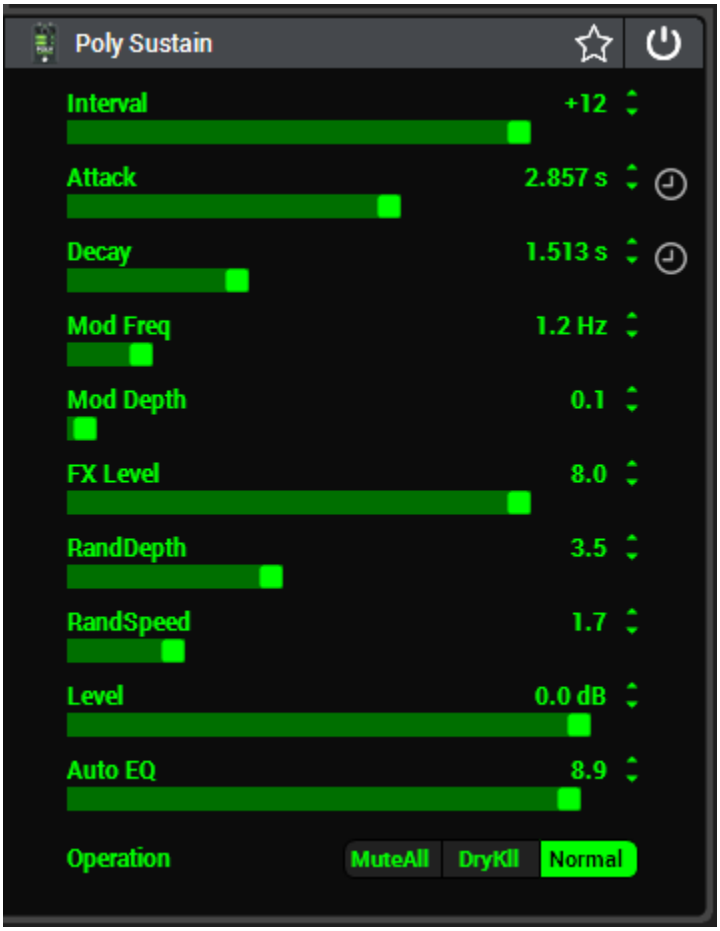
Feedback, **Mix**, **Level**, and **Trails** work the same as with other delays. The **Delay** and **Echo** modes are different. Delay repeats the same Pattern, and then fades out based on the feedback control. Echo creates echoes within the Pattern, instead of repeating the Pattern. Delay is a good choice for emphasizing the echo's rhythm, while Echo works well for "wash of sound" effects.

This variation on conventional echo is yet another reason to explore the Line 6 Original delay effects.

Poly Sustain

This mono-only effect can “freeze” your playing. It draws a lot of processing power but can inspire many creative possibilities (and not just for guitar).

To freeze audio, enable Poly Sustain (i.e., turn on its power button, or leave bypass mode) while the audio is playing. Poly Sustain locks that audio in memory. Even if you stop playing, the only way to “thaw” the frozen audio is to bypass the effect. Note that this effect isn’t about fidelity—it imparts an unusual, metallic kind of vibe. That’s part of its character.



The parameters are quite different from other Helix effects. Here’s what they do:

- **Interval** transposes the frozen audio. This real-time operation can change pitch while the frozen audio plays. The range is -36 to +24 semitones.
- **Attack** fades in the frozen audio after you enable the effect.
- **Decay** fades out the frozen audio after you bypass the effect.
- **Mod Freq** sets the rate of a vibrato-like effect.
- **Mod Depth** determines how much the vibrato will affect the audio.
- **FX Level** mixes in the frozen audio with the real-time audio, depending on the Operation switch setting (see later).
- **RandDepth** injects an unusual kind of random modulation that alters the frozen audio’s character.
- **RandSpeed** alters the random modulation’s depth.
- **Level** is the usual output level parameter.

- **Operation** has three options. *Normal* plays frozen audio (when enabled) along with the incoming audio. *DryKll* mutes the dry signal when Poly Sustain is enabled. With *Mute All*, you never hear the dry sound, but you hear Poly Sustain if it's enabled. This is useful when processing Poly Sustain in a parallel path.
- **Auto EQ** (fig. 2.60) compensates for timbral shifts when transposing up (which increases brightness) or down (which makes the sound more muffled). At 0.0, there's no compensation. 10.0 gives maximum compensation when transposing.

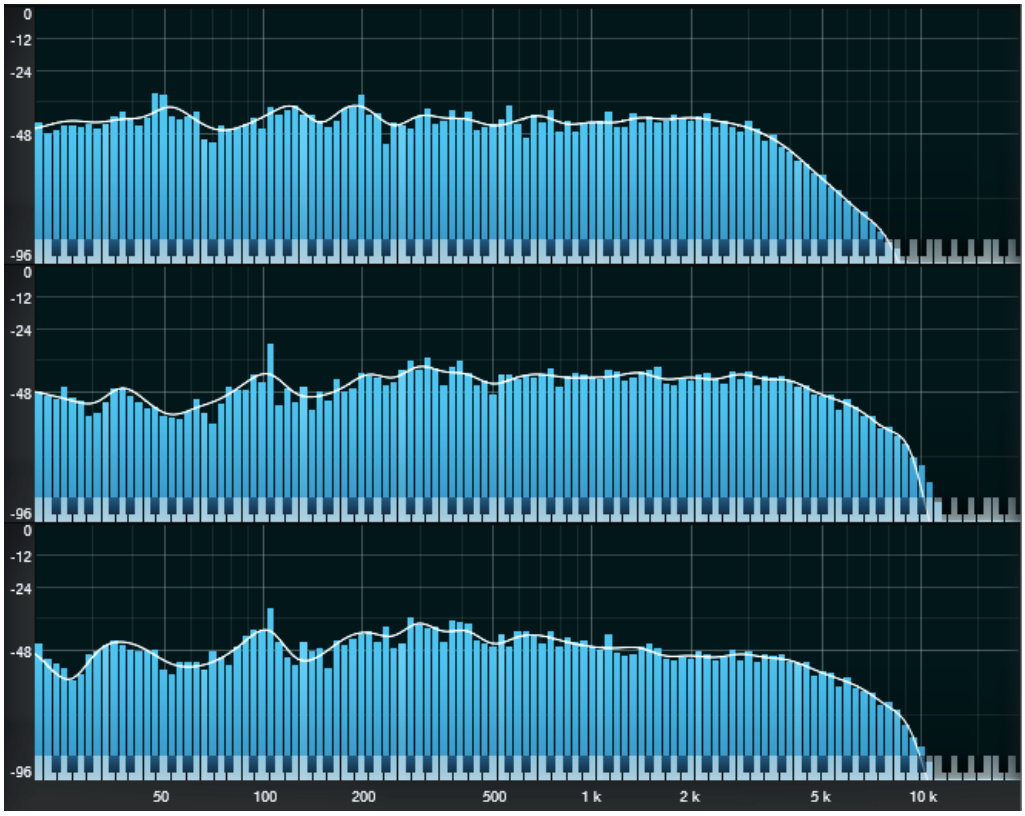


Figure 2.60 Auto EQ in action. The top image is the original waveform. The middle image transposes the original waveform up +12 semitones—note the increase in high frequencies. The lower image shows how Auto EQ reduces the high frequencies of the transposed audio to be more like the original waveform.

Glitch Delay

Some musicians do live performances—from industrial rudeness to peaceful ambience—with chaos/random-oriented pedals that rarely, if ever, produce the same results twice. You can hear an example of this in an ambient, real-time, improvised piece I did with the Line 6 HD500 (the precursor to the Helix, although the music used what are now Helix Legacy effects). Although I could do something similar again, it could never be done the same way twice.

Glitch-based music, which relies on random processes, broken or malfunctioning equipment, or pushing software to extremes, came to prominence at the end of the last century. Although it took root in avant-garde music production, elements of glitch have since become part of the mainstream…and now, part of Helix.

The goal of glitch is to produce surprising results due to unpredictability, which makes explaining how this processor works daunting. The parameters all interact in ways where you won't be able to obtain predictable results over time. However, the Glitch Delay is more versatile than the name might imply. It's capable of generating beautiful, flowing soundscapes, as well as insane sonic flurries.

Glitch Delay produces repeats from an input signal, but the similarities to conventional delay end there. These repeats are sliced into smaller divisions, which are controlled by various parameters—primarily pitch, rhythm, randomness, reversal, and shuffling their order.

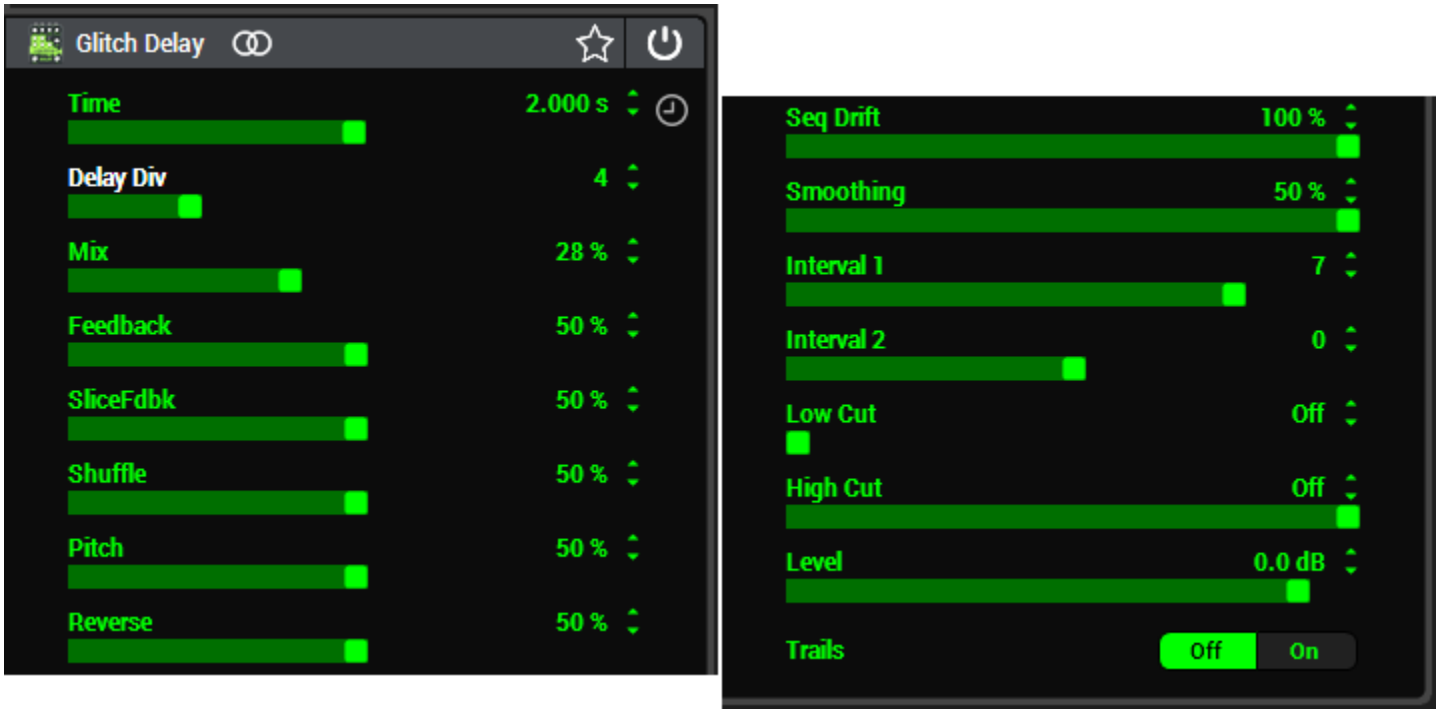


Figure 2.61 Glitch Delay parameters interact, which makes obtaining predictable results more difficult. These are the settings for the **Sweet Soundscape** preset.

Let's look at the "normal" parameters first (fig. 2.61). **Time** is the master delay parameter, where you can choose milliseconds, or sync to tempo. **Mix** determines the dry/delay balance, and **Feedback** controls the master delay feedback. Now, let's enter uncharted territories.

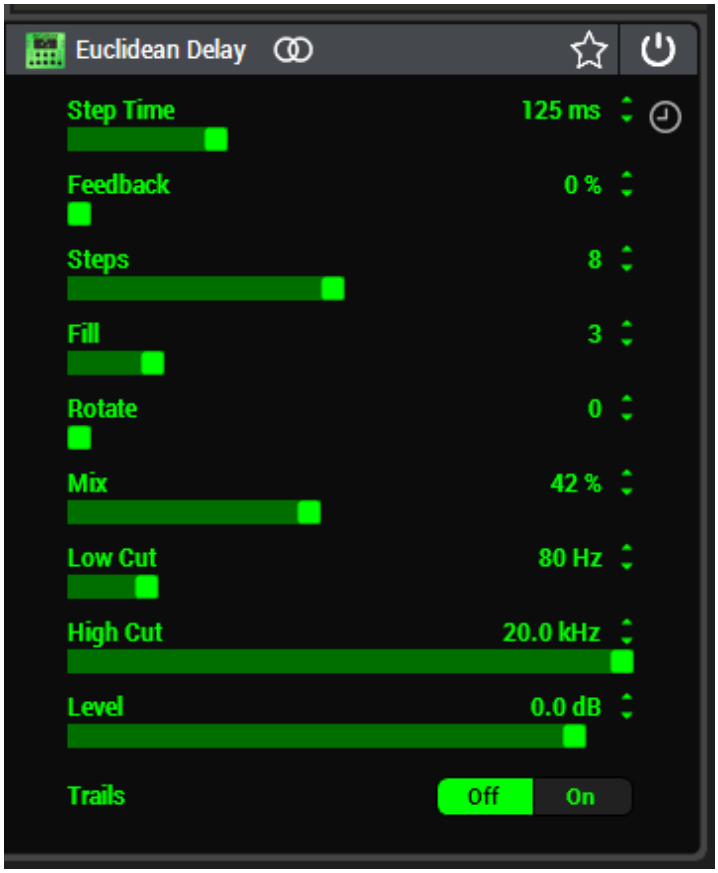
One way to wrap your head around how Glitch Delay works is to run through the following steps. Start by setting **Time** to 2.000 seconds, **Delay Div** (slices) to 2, **Mix** to 50%, **High Cut** and **Low Cut** filters to Off, and all other parameters to 0. For an audio source, use a short impulse, like a quick upstroke on a chord.

1. Vary the Div parameter. This will sound like a conventional delay, and there will be no difference among the various settings.
2. Now set Delay Div to 4, and Slice Feedback to 50%.
3. Turn up Reverse to 100%, and you'll hear that the delays are now reversed except for the initial dry delay, which will repeat. Reverse sets the *probability* that a slice will be reversed, so lower values mean that fewer slices will be reversed.
4. Return Reverse to 0%. Set Interval 1 to 12, and Interval 2 to -12.
5. When you play back, nothing happens to the Pitch. So, turn Pitch up to 100%. This parameter determines the probability that a repeated slice will change pitch, according to the setting of the two Pitch intervals (in this example, octaves). The higher the pitch percentage, the greater the odds that the slice's pitch will change. Because there's slice feedback, a pitch change will be re-applied on subsequent echoes.
6. Now let's make everything more complex (why not?). Turn up the main Feedback to 50%, then play with the Delay Div parameter after hitting one quick upstroke. You'll hear cascades of glitchy repeats. The nature of these repeats will be same, but if you play another upstroke, you'll hear a different pattern that also repeats.
7. To change this up, turn up Seq Drift. This causes the repeats to mutate, and the effect becomes more pronounced with higher Div settings. Setting this to 0 "freezes" the repeat pattern. Shuffle changes the slice order, and I often consider it as working in tandem with Seq Drift.

As mentioned previously, though, Glitch Delay can produce some beautiful sounds. Load the preset **Sweet Soundscape** from the *Free Files > Chapter 3 Presets* folder, and you'll hear what I mean.

Euclidean Delay

This is a variation on a multi-tap delay effect. It can have up to 16 taps (Steps), with a fixed delay (Step Time) between steps. To find the total delay, multiply the **Step Time** by the number of **Steps**.



- **Fill** sets the number of taps that are active. You don't choose these taps; instead, an algorithm distributes the selected number of taps as evenly as possible among the steps. The one exception is if the number of Fills equals or exceeds the number of steps, in which case the Taps provide evenly spaced echoes, like a standard delay.
- **Feedback** repeats the sequence of delays.
- **Mix** does the usual—sets the balance of dry and delayed sounds.
- Similarly to other effects, **Low Cut** and **High Cut** tailor the frequency response of the repeats. For example, by reducing the highs with feedback repeating the delays, subsequent repeats will sound progressively less bright.
- **Rotate** moves the position of all active Taps within the Steps by the same amount. Hopefully Line 6 won't mind if I do a "remix" of the image that came with the release notes (fig. 2.62), because it shows the relationship between Fills, Steps, and Rotate very clearly. Each circle represents a step. The green dots indicate active taps.

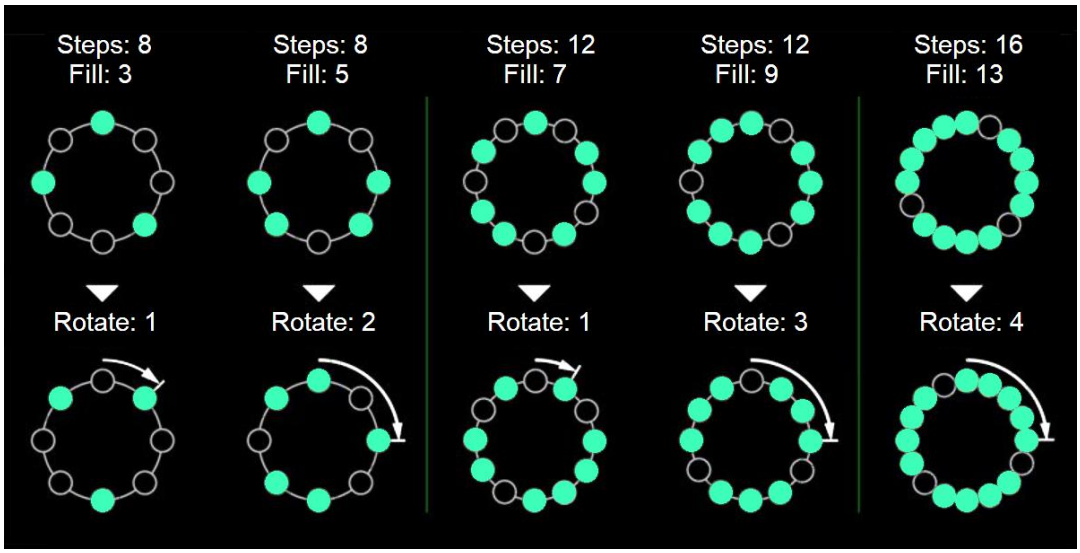


Figure 2.62 The relationship between Fill, Steps, and Rotate with respect to active taps.

Also note the Euclidian Delay is mono in, stereo out. As it goes through its steps, the stereo image starts in the center, pans toward the right, pans toward the left, then pans back until it's one step before center.

Line 6 mentions a cool application in the release notes—placing them in parallel, set to different Fills and Steps, for polyrhythmic effects. Another is taking advantage of the way it pans echoes.

Heliosphere

This Line 6 original delay inserts reverb in the delay's feedback loop, which results in a uniquely ethereal sound. Depending on the settings, it's also possible to use this as a standard delay, standard reverb, or a "morphed" combination of the two.

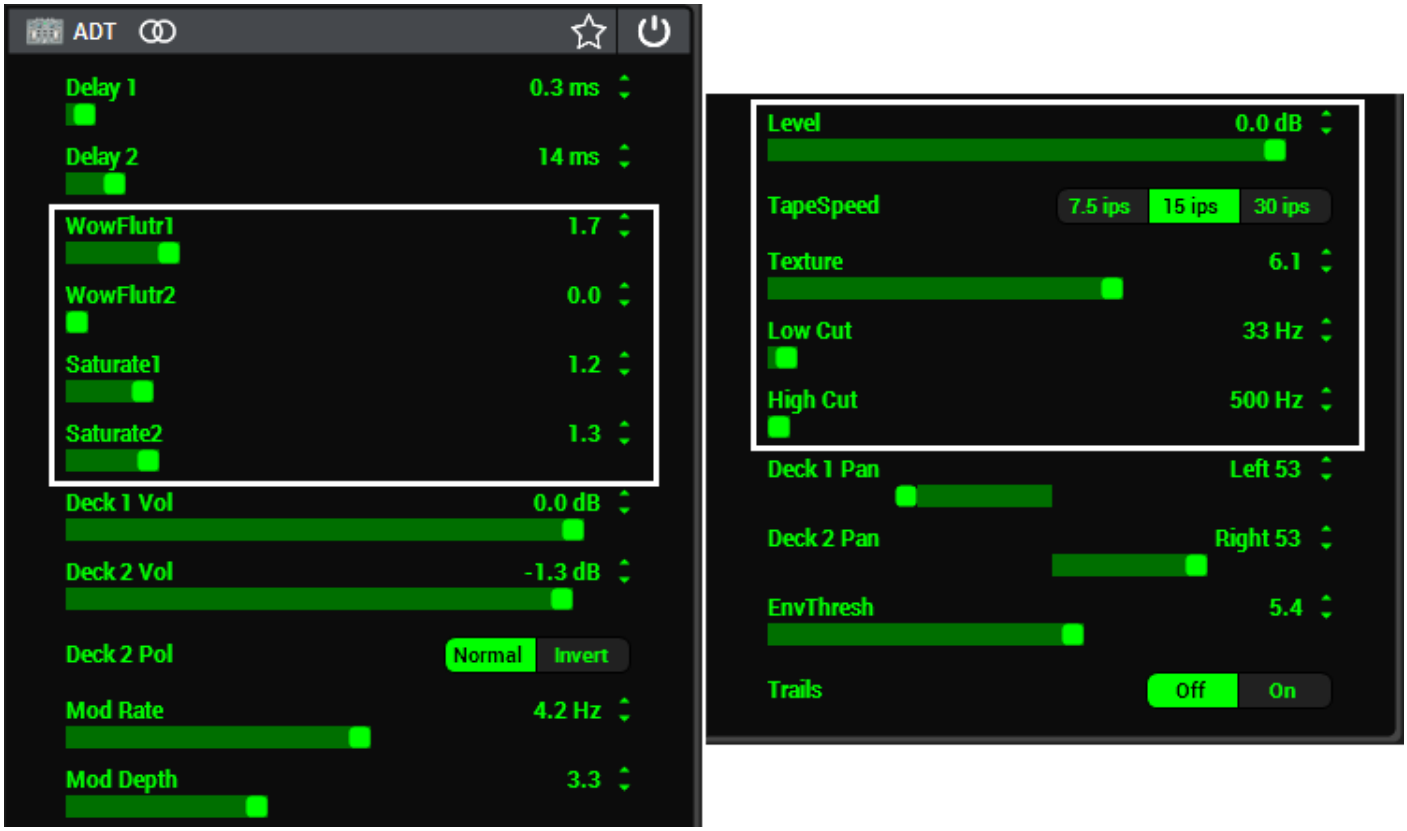
- **Time** sets the interval between delays, from 0.0 milliseconds to 8.00 seconds (or according to Note value). When set to 0.0 ms, Heliosphere sounds more like a standard reverb than its characteristic delay-meets-reverb effect. In Stereo mode, the Reverb is stereo, and there are left/right delays.
- **Feedback** sets the number of repeats. With short delay times and 100% feedback, the reverb can self-oscillate. This has some wild potential for sound design.
- **Rate** sets the speed for periodic delay time modulation. This ranges from slow speeds suitable for chorusing, to faster rates that are more like vibrato.
- **Depth** determines the amount of modulation applied to the delayed audio.
- **Mix** sets the percentage of delayed to dry sound, from 0% to 100%.
- The **Level, Scale,** and **Trails** parameters have complete descriptions in the Delay Blocks section.
- **Reverb Mix** is a secondary mix control that determines the reverb level present in the feedback loop. At 0%, the echoes are dry, with no reverb. Heliosphere then acts like a standard delay line. At 100%, the echoes consist only of the reverb effect, with no dry sound.
- **Reverb Decay** determines the reverb's decay time.
- **Headroom** has a complete description in the Modulation Blocks section.

Tip: For an "infinite reverb" effect, set Rev Decay to Maximum, as much Feedback as possible without self-oscillation, and a short enough Time so that the reverb's decay spills over into the start of the next reverberated echo. This creates a continuous, seemingly never-ending reverb.

Tip: Stressed-out? Tired? Had a tough day? Put on headphones, and then play through this for 30 minutes.

ADT

If Double Take and Retro Reel had a baby, and it grew up and went to college, it would end up as this Line 6 original effect that combines double-tracking and tape emulation. Back to the 60s…



To understand this effect's classic use case, let's review how tape-based automatic double-tracking (ADT) worked. Standard double-tracking required a musician to play the same part again. The goal was to match the original track as closely as possible. However, it was impossible to match a part exactly. The slight timing and pitch differences made the part sound bigger than if the two parts were perfect copies. Also, if panned in stereo, the two tracks created width because they were different. Had they been identical, they would have sounded like a mono track, panned to center.

An alternative to standard double-tracking involved delay. Delaying the original track, modulating the delay to give subtle timing changes, and mixing the two together produced a double-tracked effect. The Double Take block uses this approach. However, with conventional double-tracking, a player would sometimes play ahead of the original part, and sometimes behind. When using delay, the double-tracked part is always behind the original part. This sounds less authentic.

Ken Townsend of Abbey Road Studios is credited with creating true, automatic double-tracking by using two tape recorders. One of these had variable speed. Speeding up or slowing down the tape recorder with the copied track allowed it to lead or lag the original track. This sounded more like authentic double-tracking, compared to the copy always being later than the original track.

This ADT block models two tape decks in parallel. The parameters outlined in white are the same as the Retro Reel tape emulator, except that each deck has its own set of **Wowflutr** and **Saturate** parameters. Although I'm not a fan of wow and flutter, they're useful here. Adding wow and flutter creates more variations between the two double-tracked lines.

The Delay 2 time can lead *or* lag the Delay 1 time by setting Delay 2 slightly longer than Delay 1, and applying modulation to Delay 2. This produces the true effect of tape-based, automatic double-tracking.

- **Delay 1** can delay from 0.0 ms to 20 ms, and **Delay 2**, from 0.0 ms to 200 ms. Typically, Delay 1 would be fairly short, while Delay 2 would emulate the delay caused by double-tracking. Longer Delay 2 settings produce double-tracking with "looser" timing.
- **Deck 1 Vol** and **Deck 2 Vol** are independent level controls for each deck.
- **Deck 1 Pan** and **Deck 2 Pan** are available in Stereo mode. They position each deck in the stereo field.
- **Deck 2 Pol** switches deck 2's polarity from positive to negative. When negative, and the double-tracked lines "cross over" each other so that they have the same delay time for an instant, there's some of the through-zero flanging vibe that was popular back in the 60s.
- **Mod Rate** and **Mod Depth** apply periodic, sine wave frequency modulation to Deck 2. Rate sets the modulation speed, while Depth applies the amount of modulation.
- **Envelope,** when on, correlates the Delay 2 delay time to your playing. Picking harder (see EnvThresh) subtly alters Delay 2's delay time. This adds slight pitch fluctuations that enhance the double-tracking effect.
- **EnvThresh** sets the level your playing must exceed to trigger the envelope.

Crisscross

This Line 6 Original effect incorporates two delay lines (Delay Line A and Delay Line B), with cross-feedback between them. Having one delay line feed into the other, and vice-versa, allows a more intricate delay effect than a standard dual delay. The **Sample Rate** and **Bit Depth** parameters emulate the lower-fi sound associated with vintage digital delays. To save on memory costs by reducing the storage space needed for samples, these older delays used slower clock speeds and reduced bit depth.

The screenshot splits the effect into two parts. The left section shows standard delay parameters. The right side includes modulation, EQ, digital fidelity, and other parameters.

The Delay Blocks section includes descriptions for the **Mix**, **Level**, **Headroom**, **High Cut**, **Low Cut**, and **Trails** parameters. With Crisscross, these are common controls that affect both delay lines.

- **Time A** and **Time B** set the delay time for the A and B lines. Note that unlike most other Helix delays, there's no Scale parameter. So, changing tempo doesn't maintain the time relationship of the two lines.
- **Feedback A** and **Feedback B** send audio back from their respective delay lines to their respective delay line inputs, like a standard delay. 0% produces a single delay, while 100% gives infinite echoes. The echoes stop when Helix loses power, or the sun goes supernova.
- **Pan A** and **Pan B** position their respective delay lines in the stereo field.

Tip: Panning one delay hard left or right, and one delay to center, leaves some space in the channel opposite to the one with the delay. This can be an effective mixing technique for LCR mixes.

- **Crossfeed** is a common control for both delay lines. It determines the level of the audio from Delay A that enters Delay B, and likewise, the level of the audio from Delay B that enters Delay A. Because each delay is continuously feeding the other delay, audio travels back and forth between the two delays, and is influenced further by the Feedback parameters for the individual delay lines.

- **Mod Rate** controls the speed of a low-frequency oscillator that modulates the delay time. Although modulating any delay can give chorusing effects, Crisscross can combine crossfeeding with feedback to accentuate chorusing's lush qualities.
- **Mod Depth** sets the amount of modulation applied to the two delay lines.
- **Shape** selects the modulation waveform (triangle or sine wave).
- **Phase** offsets the modulation waveform for the two delay lines (fig. 2.63).

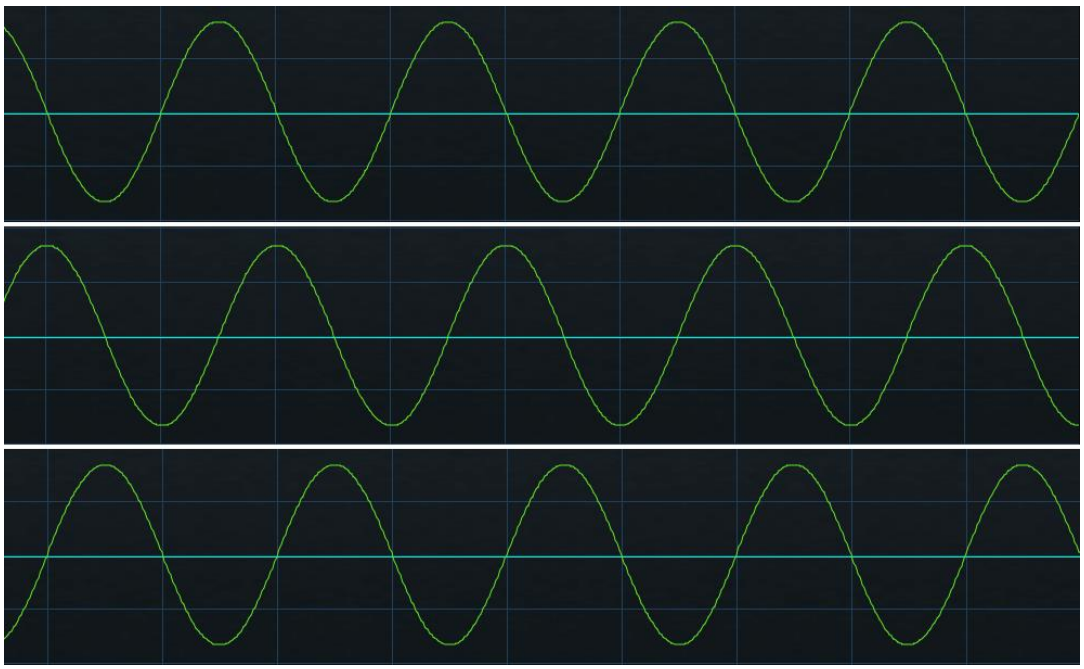


Figure 2.63 Phase offsets the waveform that modulates the two delay lines. Top: 0 degrees (in-phase). Middle: 90 degrees out of phase. Bottom: 180 degrees out of phase, where the peaks and valleys are reversed compared to the in-phase waveform.

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- **Sample Rate** influences the high frequency response. The higher the sampling rate, the higher the frequency response (fig. 2.64).

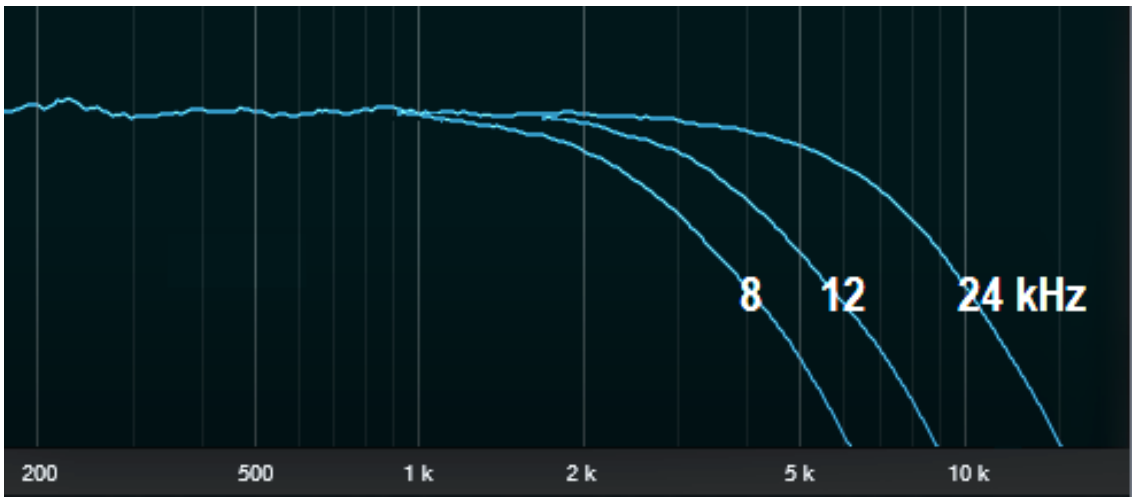


Figure 2.64 Frequency response curves for three different sample rates. 44.1 and 48 kHz are much flatter.

- **Bit Depth** sets the audio resolution for echoes. 24 bits is the standard for today's high-resolution audio. 16 bits is the same resolution as a CD. Older delay lines often used 12- or even 10-bit resolution.

Anything lower than that is best used for special effects, toys with digital audio, and lo-fi, although some (very) early digital delays used 8-bit resolution.

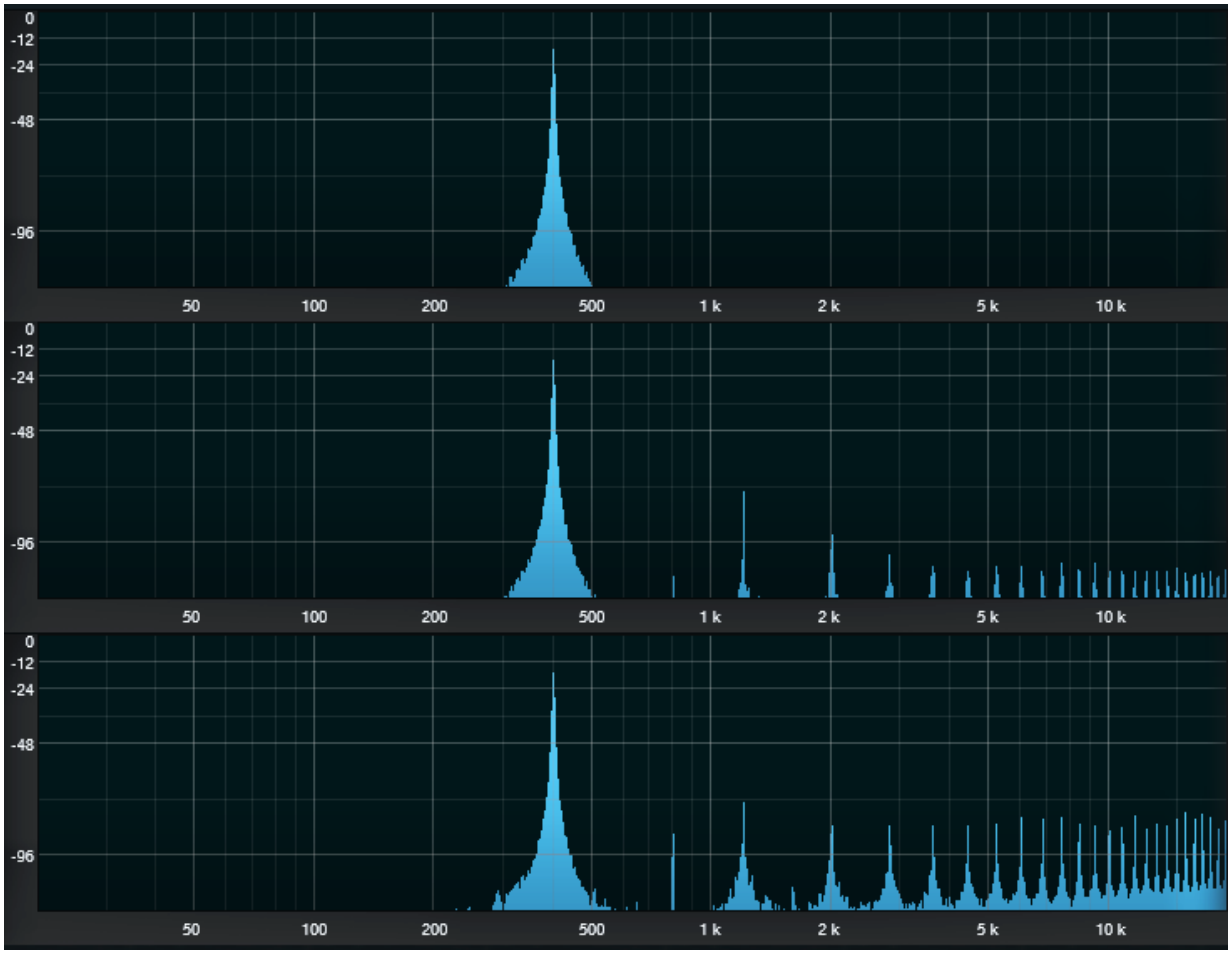
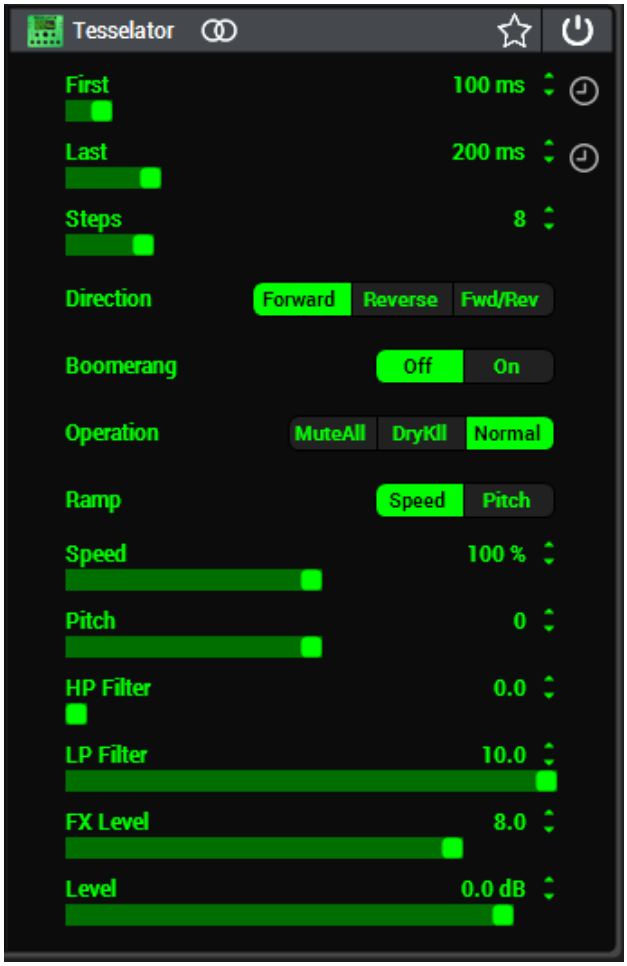


Figure 2.65 Lower bit resolutions increase the amount of distortion components, which are the peaks to the left of the main 400 Hz sine wave. Top: 24-bit resolution. Middle: 12-bit resolution. Bottom: 10-bit resolution.

Fig. 2.65 shows three different bit resolutions. The sine wave test tone is 400 Hz. With 24-bit resolution, there's no harmonic distortion—just the tone itself. With 12-bit resolution, you can see harmonic distortion components creeping in. Bear in mind, though, these are still around -70 dB below the test tone, so they're not very noticeable. 10-bit resolution adds considerably more distortion, but again, the real test is in the listening. It's not horrible, just gritty—which may be the sonic character you want.

Tesselator

This Line 6 Original settles the question of whether alien technology has been reverse-engineered. The release notes describe it as “part morphing delay, part loop sampler, part drone machine...it’s stellar for creating rhythmic pads, textures, or pitch/filter ramp effects to play over.” That’s it in a nutshell.



Like the OSC and some other blocks, Tesselator defaults to bypassed. Enabling it captures audio that occurred *before* you enabled Tesselator, so it’s best to assign this block to a footswitch. Tesselator separates the captured audio into two parts, called *first* and *last*, which are *not* the first and last parts of the captured audio. Instead, the captured audio’s beginning is copied to two parts. One of these is the first audio segment in a sequence of events. The other is the last audio segment in a sequence of events. Each event is a *step* in the sequence.

Tesselator’s main talent is being able to modify what occurs between the first and last steps, with parameters that are eligible for real-time editing. This effect just about begs for adding one (or two!) more expression pedals, so you can vary parameters in real time. Edits affect the tessellated sound more with each step, until reaching the target setting at the last step…which may return to the start…or go into reverse…or boomerang back…or…well, keep reading. Fig. 2.66 shows multiple graphic examples of how tessellation affects audio, depending on particular settings.

- **First** sets the capture time for the audio that occurred *prior* to enabling Tesselator. This becomes the first step in the sequence of events. The time can be in ms (up to 1.00 second) or note values. When choosing note values, there’s sufficient memory to capture a full measure at tempos down to 60 bpm.

- **Last** determines the length of the sequence's last event. If the Last time is shorter than the First time, the repeated steps become shorter as the sequence progresses. If the Last time is longer than the First time, then the steps lengthen as the sequence progresses. If the First and Last times are the same, the step length remains constant (Ex. 1 in fig. 2.66 below).
- **Steps** chooses the number of steps in the sequence, from 1 to 50. For example, if the first step is 100 ms and the last step is 500 ms, each successive step in the sequence will lengthen progressively, from 100 ms to 500 ms. The more steps you have, the longer it takes to reach the last step and therefore, the longer it takes to alter the sequence's characteristics.
- **Direction** determines how each step plays back. *Forward* plays back each step normally, from the step's start to the end (Ex. 2a below). *Reverse* plays back each step in reverse, from the step's end back to the start (Ex. 2b below). *PingPong* alternates steps between forward and reverse playback (Ex. 2c below)
- **Boomerang** alters how the last step is handled. When *off*, the sequence's last step repeats indefinitely. When *on*, after all steps play forward from first to last, the steps then play in reverse order from last to first, then forward again, and so on. Note that when the steps play in reverse order, the audio itself doesn't play back in reverse. This affects only the *order* of the steps (Ex. 3a below).
- **Operation** has three options that determine how the Tesselator interacts with a preset when it's first enabled (remember, it's bypassed by default). *Normal* plays the tesselated audio along with the dry, incoming audio. *DryKll* mutes the dry signal, so you hear only tessellated audio. With *Mute All*, you never hear the dry sound, even with Tesselator off. However, you will hear the tesselated sound when enabled. This is useful when the Tesselator is in a parallel path.

Tip: With Tesselator in a parallel path, you may want to leave the block enabled, yet be able to bring the tessellated sound in and out of the path. To do this, assign a second footswitch to toggle between Mute All (dry sound, but no tessellated sound) and Dry Kill (no dry sound, but tessellated sound).

- **Ramp** chooses either the Speed or Pitch slider below the Speed/Pitch type selection. These parameters determine whether any playback speed or pitch changes that ramp across the sequence reference a percentage or semitone change.
- **Speed** sets the target playback speed of the last step, from 0% speed (stopped) to 200% speed. The speed changes are distributed over the sequence's length. For example, a setting of 200% causes the last step to play back twice as fast as the first step, thus playing it an octave higher. However, note this doesn't change the steps' durations. Think of "speed" like changing a keyboard's note pitch higher or lower with a pitch bend wheel.
- **Pitch** edits the target pitch of the last step, from -12 to +12 semitones (Ex. 3c below) For example, with a pitch setting of -12, the last step will be an octave lower than the first step. Think of this like changing a keyboard's note pitch by playing a different key on the keyboard.
- **HP Filter** is not like a conventional low cut (highpass) filter. Its settings choose the filter frequency for the last step. For example, higher values will filter out progressively more bass, until reaching the target cutoff on the sequence's last step.
- **LP Filter** is not like a conventional high cut (lowpass) filter. Like the HP Filter, its settings choose the filter frequency for the last step. For example, higher values will filter out progressively more treble until the sequence's last step (Ex. 3b below).
- **FX Level** controls the processed signal's mix with a dry signal (if present, as selected by **Operation**).
- **Level** sets the block's overall output, which is useful for gain-staging.

Tesselator Examples

Tesselator can create hundreds of unique sounds. Fig. 2.66 shows just a few examples to help get you started on discovering what this novel effect can do.

Example 1

If the First and Last values are the same (e.g., 1/4 note), the same length of audio repeats until you bypass Tesselator. This example acts very much like the Ratchet effect (see the description in the next effect block), except Tesselator captures the audio *before* the stomp press, not after.

Example 2

If the Last value is shorter than the First value, the sequence's steps become progressively shorter (Ex. 2a). If the Last value is longer than the First value, the sequence's steps become progressively longer. The last step repeats indefinitely until you bypass Tesselator. Setting Direction to *Reverse* (Ex. 2b) reverses all steps. Setting Direction to *PingPong* (Ex. 2c) alternates between forward and reversed steps.

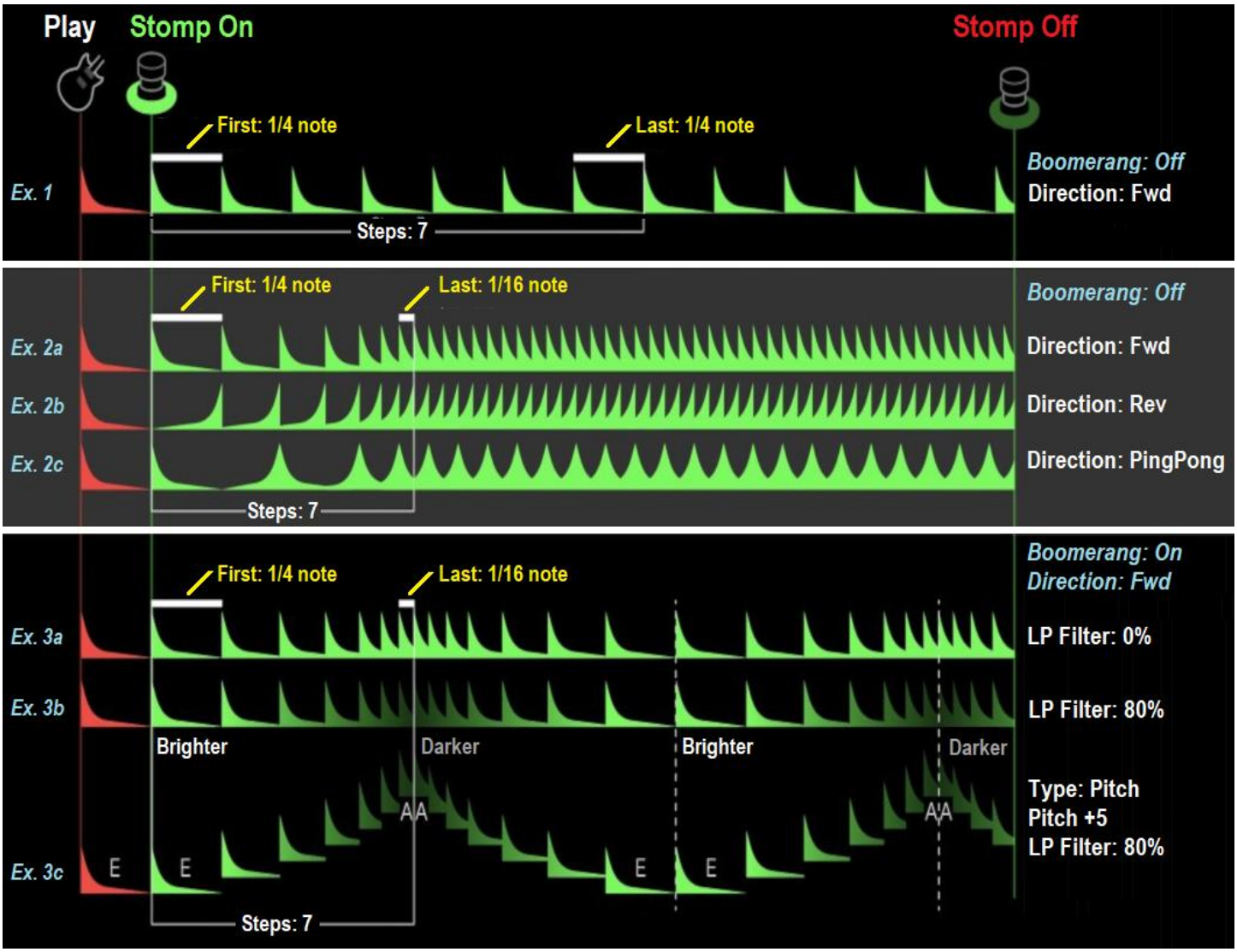


Figure 2.66 Several examples of how Tesselator processes audio, depending on different settings.

Example 3

Turning Boomerang *On* plays the entire step sequence forward, then backward, then forward again, ad infinitum (Ex. 3a) Increasing **LP Filter** to a higher value progressively reduces high frequencies for each step in the

sequence (Ex. 3b). Increasing **HP Filter** to a higher value progressively reduces low frequencies for each step in the sequence.

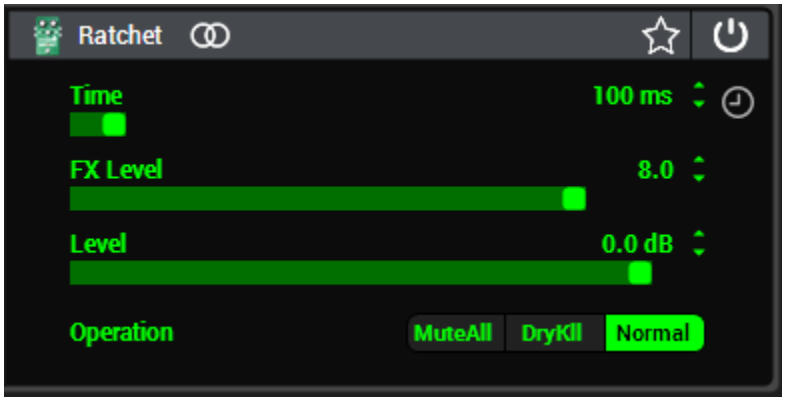
Setting **Type** to *Pitch*, and Pitch to a value other than 0.0, will change each step's pitch until it lands on the target pitch at the last step. For example, if the Pitch value is +5 (i.e., semitones) and you play an E note, the last note in the sequence will be an A, or 5 steps higher (Ex. 3c). To have the last A note repeat indefinitely instead of stepping back down to E, return Boomerang to *Off*.

Tip: Being able to change the Tesselator parameters in real time multiplies the kind of interactions it can have with other effects. For example, send it into Pitch > Dual Pitch and Reverb > Shimmer…then prepare to spend hours of bliss in droneland.

Tech Talk: What Is Tesselation?

The process of tesselation covers a surface with tiles, and there are no gaps or overlaps among them. A Roman mosaic is a good example of non-periodic tessellation, because there are no specific, repeating patterns. A periodic tessellation could be something like the hexagonal tiles that make up a bathroom's floor. A honeycomb's hexagonal cells are a natural example of periodic tesselation. The term makes sense here, because each repeat follows consecutively, with no gaps or overlaps, and covers a specific duration of sound.

Ratchet



This Line 6 Original effect captures a short piece of audio for looping. Like Tesselator, Ratchet defaults to being bypassed. Unlike Tesselator, instead of capturing audio played *prior* to being enabled, it captures audio played *after* being enabled. After capture, the audio loops repeatedly repeatedly repeatedly repeatedly until bypassed. You'll want to assign this block to a footswitch, so you can hit it to capture audio.

Tip: A momentary footswitch allows switch on and off during a solo to have bursts of repetitive audio, or on and off when playing chords for a sustain pedal-like effect.

- **Time** opens a window for capturing audio up to 1 second long. For example, suppose you want a stuttering effect. After enabling Ratchet, pause for a fraction of second before playing. Then Ratchet will loop the silence and what you played, with the loop duration equal to the Time setting. For an effect that sustains a chord, play the chord, and then enable Ratchet. This will capture the chord for the duration set by Time, and then loop it.
- **FX Level** controls the processed signal's mix with a dry signal (if present, as selected by **Operation**).
- **Level** sets the overall block's overall output, which is useful for gain-staging.
- **Operation** has three options that determine how Ratchet interacts with a preset when it's first enabled (remember, it's bypassed by default). *Normal* plays the Ratcheted audio along with the dry, incoming audio. *DryKll* mutes the dry signal, so you hear only the Rachet effect's audio. With *Mute All*, you never hear the dry sound, even with Ratchet off. However, you will hear the Ratchet sound as soon as you enable it and play something. Mute All is useful when Rachet is in a parallel path.

Reverb Blocks

Reverb is about adding ambiance, so the different models generate different types of ambient effects. However, the blocks share several common parameters:

- **Decay** sets the overall time for the reverb to decay to inaudibility. The Dynamic Hall, Dynamic Plate, and Dynamic Bloom reverbs have an Infinite decay setting. It's not really infinite, but it does last a really looooooooooooong time.
- **Predelay** introduces a short delay before the main reverb tail. This emulates the time between when a sound occurs, and when it first starts hitting reflective surfaces. Longer predelay settings make the room seem bigger. Perhaps more importantly, it keeps reverb from interfering with note attacks.
- **Mix** sets the balance of dry to reverberated sound.
- **Low Cut** reduces low frequency response starting at a specific frequency, from 20 to 500 Hz. A setting below 20 Hz bypasses the Low Cut Filter.
- **High Cut** reduces high frequency response starting at a specific frequency, from 500 Hz to 20 kHz. A setting above 20 kHz bypasses the High Cut Filter.
- **Level** sets the block's output level, which simplifies gain-staging.
- **Trails** is as important a parameter with reverb as it is with delay effects. When on, any remaining reverb continues after bypassing an effect or choosing a different Snapshot. With Trails off, bypassing an effect mutes any reverb that's still decaying.

Several of the descriptions here include 3D spectrogram images, which show the decay characteristics visually. Most images show the reverb decay diagonally (fig. 2.67), to best display the decay's character. Of course, you'll hear the difference with your ears, but the visuals help make a better connection between parameter settings, and how the reverb behaves.

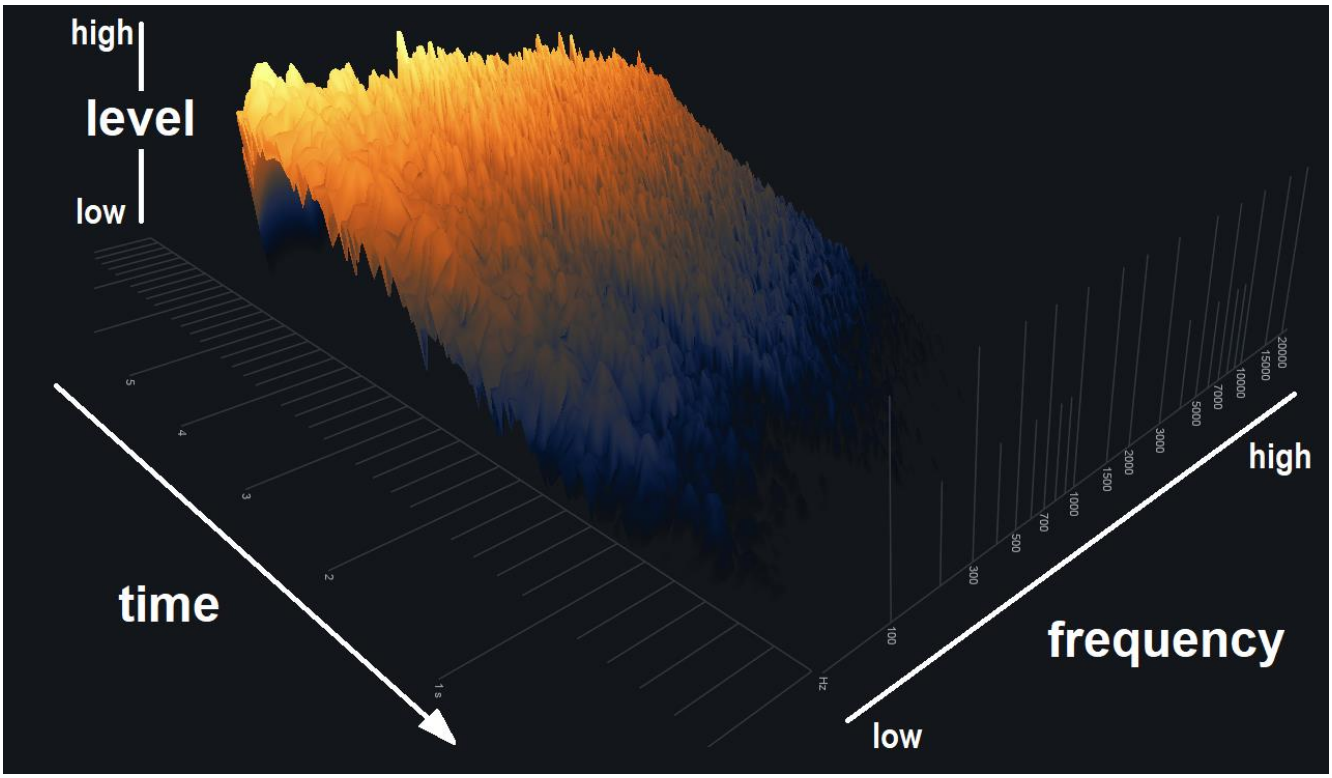


Figure 2.67 Typical 3D spectrogram of reverb decay. Brighter colors are higher levels, darker colors are lower levels. The reverb tail's "spikiness" gives a good indication of whether the reverb is smooth, or more "granular."

Applying Reverb

The "best" reverb? Reverb sounds are subjective. Many people on the web who suggest reverb settings often don't mention their playing style or intended musical goal. For example, adding a small room sound to a rockabilly guitar requires a different reverb (or at least different reverb settings) than adding a wash of reverb sound to an ambient looped-guitar solo act.

Tweaking reverbs is important. This doesn't apply to only the reverb settings themselves. Reverbs live in an ecosystem with other instruments, because their decay can spill over into what other instruments are playing. I saw a comment on a forum from someone who liked almost everything about a reverb, but didn't use it because he thought the sound was too "midrangy." The solution in that case isn't to try a different reverb first, but to put an EQ with a midrange cut before (or after) the reverb. Then, decide if that solves the issue.

Overcoming limitations. Some reverb limitations are inherent—like flutter, or other periodic variations. These interfere with a smooth, steady decay. One solution is modulating the reverb decay time somewhat, which varies the periodic changes. However, modulation can sometimes cause a different set of issues. Fortunately, algorithms continue to improve (the Dynamic Hall can hold its own with the best of them). Often, the way to overcome a limitation is to choose the right reverb for the right job. To do that, you need to know each reverb's strengths and limitations.

Take advantage of the virtual world. With Helix, you can place reverbs in series, in parallel with other tracks, preceded or followed by effects, and so much more. The "reverb" sound that rocks your world may not be an isolated reverb sound, but an effects chain that includes reverb.

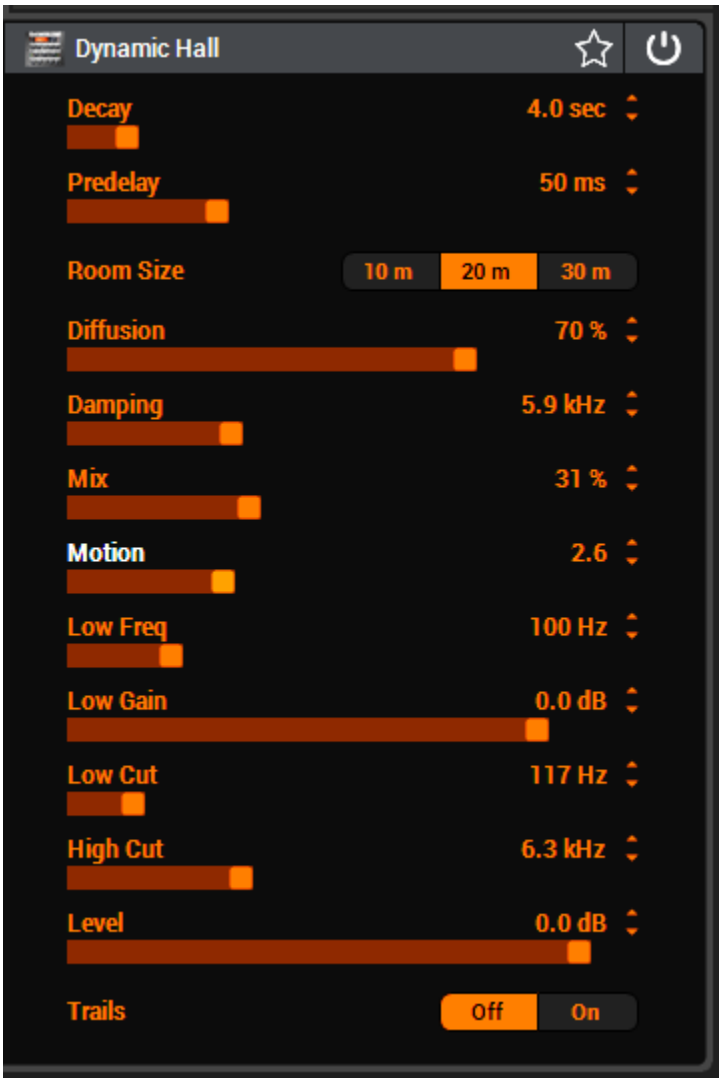
Variety is the spice of…indecision. In recording's early days, reverb was an acoustic space with one, and only one, sound. You could change the speaker sending sound into the room, or the mic that picked it up, but that was it. With Helix, understand that the different reverbs aren't all intended for you. If you want a lifelike acoustic reverb sound, the Dynamic Reverb is ideal. In a sense, it's like traditional reverb: it does one thing, really well. But if you're a cover band doing a surf song, then the Hot Springs is the ticket…or Plateaux if you're doing ambient.

The bottom line is that when evaluating the Helix reverbs (and don't forget the Legacy ones), the "best" reverb will be a moving target, because you'll have different needs for different music. Think of reverbs as different shoes—they all have the same goal of protecting your feet but can serve many different purposes. Choose reverb based on how well it fits your music.

Reality check. I understand why people obsess over reverb sounds: a beautiful, smooth, evocative reverb effect can be a total joy. But remember, in the studio, reverb is—almost by definition—in the back of the mix. And for live performance, the venue where you're playing is adding the reverb, which often makes artificial reverb unwelcome (or at least redundant). Reverb exists to support, not dominate. Spend the time to try all the Helix reverbs, so you can become familiar with their characteristics. Then, you can choose the right tool for the right musical context.

Dynamic Hall

This studio-quality reverb has the most “acoustic” sound of the currently available Helix reverbs. The Decay time can be up to 45 seconds, after which there’s an option to choose Infinity. Unless you’re adjusting other parameters to shape the Decay (see below), the attention span required to follow the reverb’s Infinity decay setting into nothingness will probably be shorter than the reverb time itself.



In addition to the parameters covered in the Reverb Blocks section (**Decay**, **Predelay**, **Mix**, **Low Cut**, **High Cut**, **Level**, and **Trails**), the Dynamic Hall includes:

- **Predelay** introduces a short delay before the main reverb tail. This emulates the time between when a sound occurs, and when it first starts hitting reflective surfaces. Longer predelay settings make the room seem bigger. Perhaps more importantly, it keeps the reverb from interfering with note attacks.
- **Room Size.** Different rooms reverberate differently, so this parameter chooses among three room algorithms. Smaller spaces build up their reverb more quickly—like a faster attack. The reverb in bigger rooms “blooms” more, and the attack takes longer to develop.
- **Diffusion.** Low diffusion settings spread echoes further apart, which applies echo more sparingly to the audio. Sustained sounds often use low diffusion settings. Higher settings pack echoes closer together, which creates a smoother echo effect with percussive sounds. For example, low diffusion is common with vocals, and high diffusion with percussion.

- **Damping.** In rooms with soft surfaces, like a hall filled with people, high frequencies decay faster than low frequencies. In rooms with hard surfaces, high frequencies bounce around the room longer. Damping is like a lowpass filter, whose cutoff goes lower as the reverb fades out (fig. 2.68).

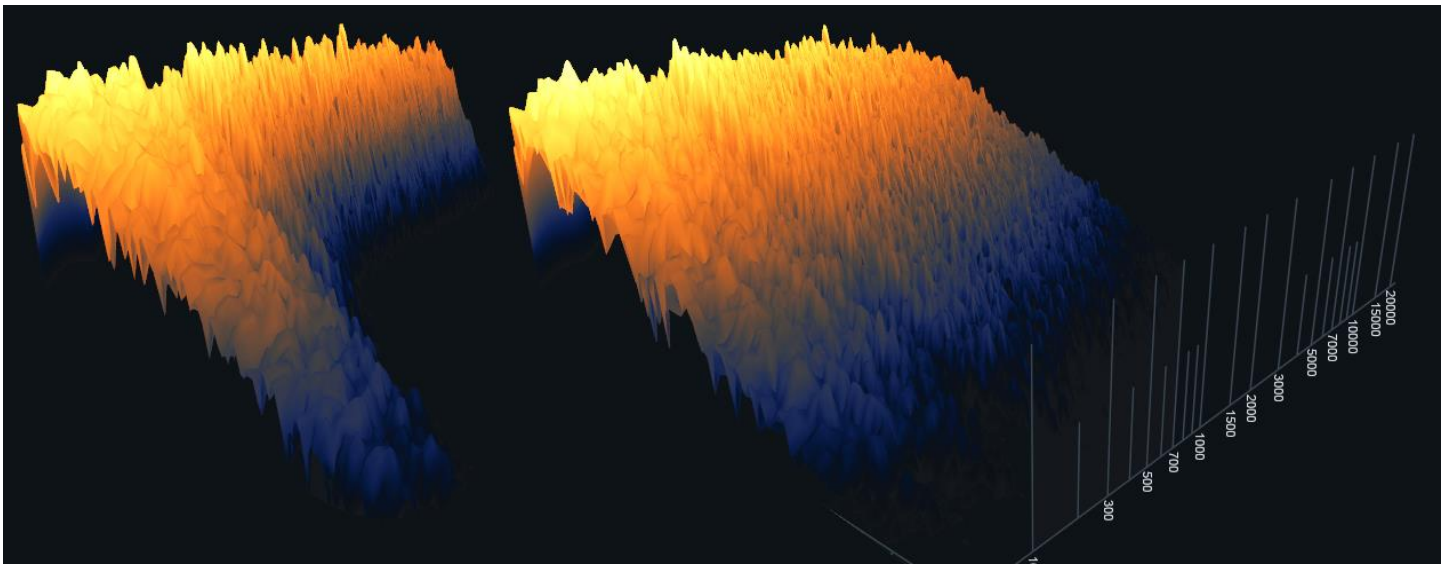
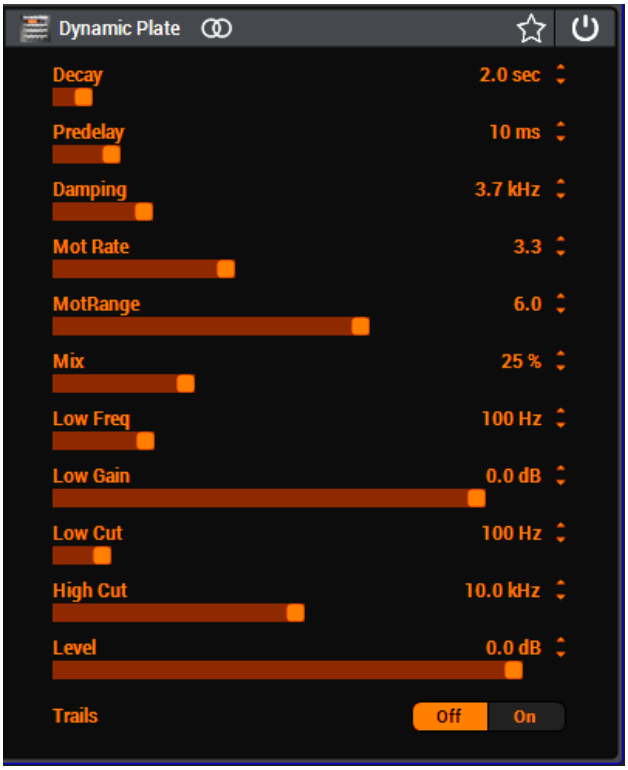


Figure 2.68 Although the Decay parameters are the same, the reverb on the left has lots of damping, so the high frequencies roll off faster than the reverb on the right, which has minimal damping. The lower frequencies are relatively unaffected.

- **Motion** is like a mini-chorus that adds subtle, randomized variations to the reverb. This can reduce "metallic" artifacts that sometimes happen with static delay times. Higher values are more noticeable as modulation, so more is not necessarily better.
- **Low Freq.** The Dynamic Hall includes a crossover that separates the high and low frequencies into two separate bands. This parameter sets the dividing line for the low-frequency range, which the Low Gain parameter (see next) affects. Too much low-end reverb can muddy the sound. (On the bright side, it can also give a kick drum the coveted *Kick Drum of Doom* effect.)
- **Low Gain.** When set to 0.0, low and high frequencies decay at the same rate. Values below 0.0 cause low frequencies to decay faster than high frequencies. Values above 0.0 cause low frequencies to decay more slowly than high frequencies.

Dynamic Plate

Dynamic Hall was the first Helix reverb to emulate high-end recording studio reverbs. Dynamic Plate complements this with a high-end, mechanical plate reverb. In addition to the parameters covered in the Reverb Blocks section (**Decay**, **Predelay**, **Mix**, **Low Cut**, **High Cut**, **Level**, and **Trails**), the Dynamic Plate includes:



- **Predelay** introduces a short delay before the main reverb tail. This emulates the time between when a sound occurs, and when it first starts hitting reflective surfaces. Longer predelay settings make the room seem bigger. Perhaps more importantly, it prevents the reverb from interfering with note attacks.
- **Damping.** Plate reverbs used damping plates, controlled by servo motors, to vary the decay time and influence the high-frequency response. As with the Dynamic Hall effect, damping acts like a lowpass filter, whose cutoff goes lower as the reverb fades out (fig. 2.69).

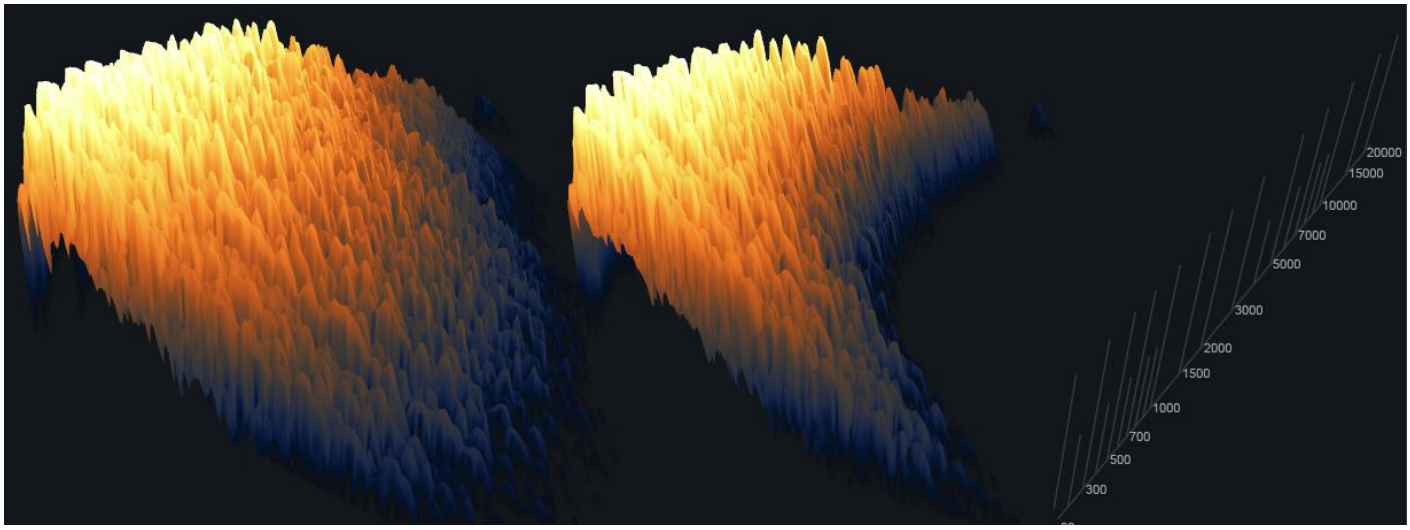


Figure 2.69 The left reverb tail has no damping. The right reverb tail's damping is set to 1 kHz, which is fairly extreme. Note how quickly the high frequencies decay.

- **MotRate** and **MotRange** work together. Plate reverbs had more resonances than acoustic reverbs, and were more subject to changes in plate tension or temperature. **MotRate** sets how rapidly the echoes' intensity changes due to these types of variations, while **MotRange** determines how much of the reverb these resonances affect (fig. 2.70). This is like the Modulation parameter in other reverbs.

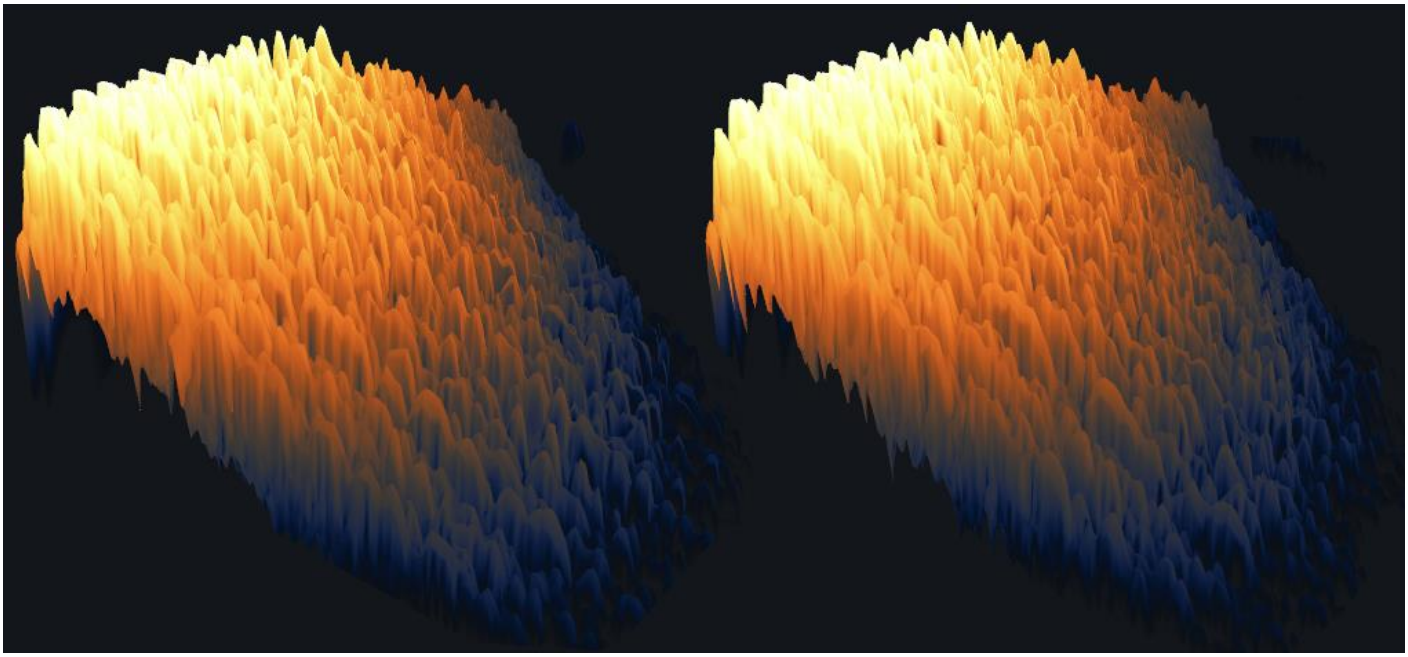


Figure 2.70 The left reverb tail has **MotRate** and **MotRange** set to minimum. The right reverb tail has both controls set to maximum. Note how the right reverb tail is more complex and varied, especially in the midrange.

- **Low Freq.** A crossover separates the high and low frequencies into two separate bands. This parameter sets the dividing line for the low-frequency range, which the Low Gain parameter (see next) affects. Too much low-end reverb can muddy the sound.
- **Low Gain.** When set to 0.0, low and high frequencies decay at the same rate. Values below 0.0 cause low frequencies to decay faster than high frequencies. Values above 0.0 cause low frequencies to decay more slowly than high frequencies.

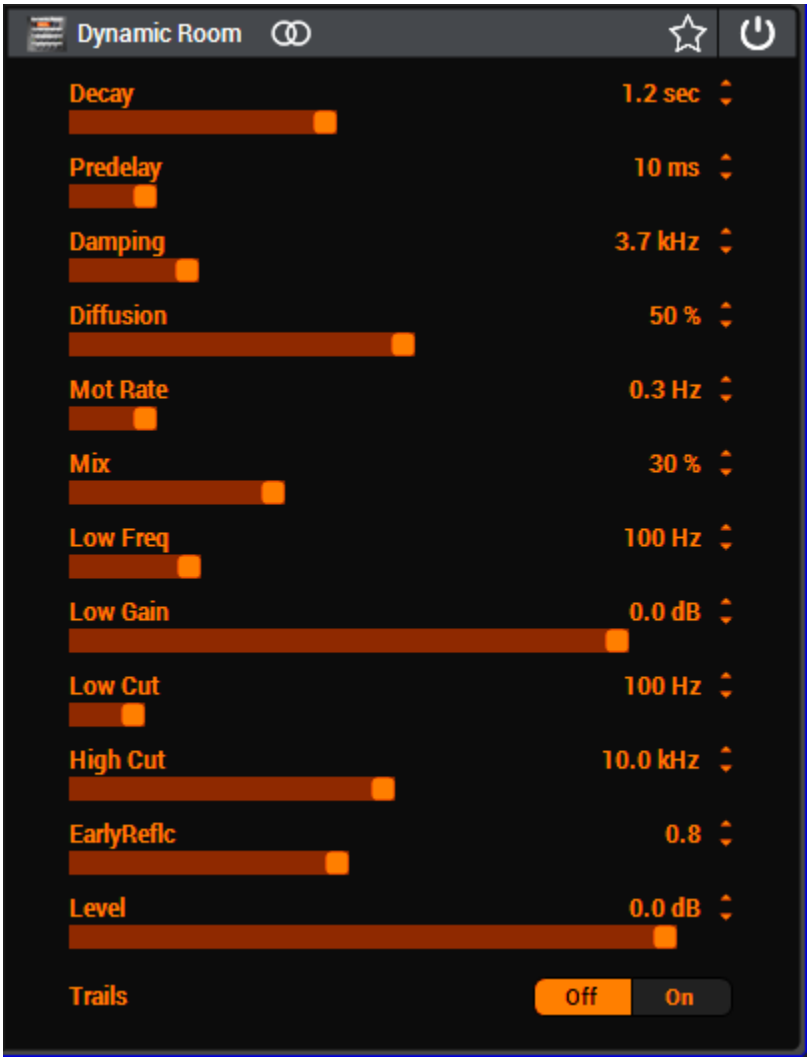
Plate Reverb Tips

Different reverbs have different characteristics, so choose the reverb that best matches how you want the track to sound.

- Plate reverbs have a shorter pre-delay than acoustic reverbs, and they also have fewer early reflections. So, they're seldom used to define a space in the same way as a Hall or Room reverb.
- Traditionally, plate reverbs initially have a brighter sound than acoustic spaces. To emphasize this, reduce the low gain so that the low frequencies decay faster than the high frequencies.
- You can reduce a plate reverb's brightness by increasing the damping.
- The Motion parameters add a "secret sauce" that can make the reverb stand out more than reverbs that emulate an acoustic space.

Dynamic Room

This completes Helix's studio-level reverb "power trio" of high-end studio processors: Dynamic Hall, Dynamic Plate, and Dynamic Room.



The parameters are optimized to emulate smaller, yet complex, spaces. In addition to the parameters covered in the Reverb Blocks section (**Decay**, **Predelay**, **Mix**, **Low Cut**, **High Cut**, **Level**, and **Trails**), the Dynamic Plate also includes:

- **Damping** sets the rate at which high frequencies decay in a room. The rate of high-frequency decay depends on whether the room has hard, reflective surfaces (less damping, so high frequencies take longer to decay) or soft, absorptive surfaces (more damping, so high frequencies decay more rapidly). For example, a small rehearsal room with wood walls would have less damping than a room with carpeting and curtains.
- **Diffusion** spaces discrete echoes closer together (fig. 2.71). Percussive sounds generally use more diffusion to minimize obvious, discrete echoes—the "marbles bouncing on a steel plate" kind of reverb effect. For sustained sounds (like vocals), less diffusion gives a sparser reverb, which opens up more space for the sustained sound.

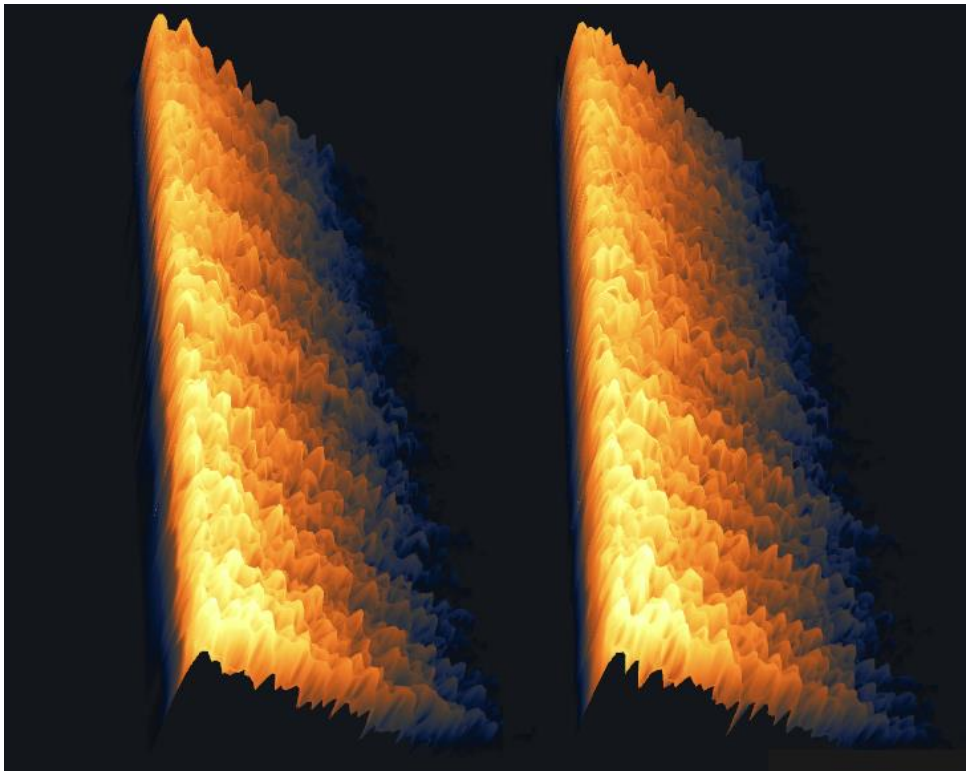


Figure 2.71 Low diffusion is on the left, and high diffusion on the right. Compare the two tails. With high diffusion, the tail's "bumps" (echoes) are smaller, less jagged, and packed closer together.

- **MotRate** modulates the reverberation to vary the reverb characteristics. Most acoustic spaces have some variation over time, even if it's due only to moving your head around and picking up different reflections in different positions, or people moving around in the space.
- **Low Freq.** A crossover separates the high and low frequencies into two separate bands. This parameter sets the dividing line for the low-frequency range, which the Low Gain parameter (see next) affects. Too much low-end reverb gives a booming, dark sound.
- **Low Gain.** When set to 0.0, low and high frequencies decay at the same rate. Values below 0.0 cause low frequencies to decay faster than high frequencies. Values above 0.0 cause low frequencies to decay more slowly than high frequencies.

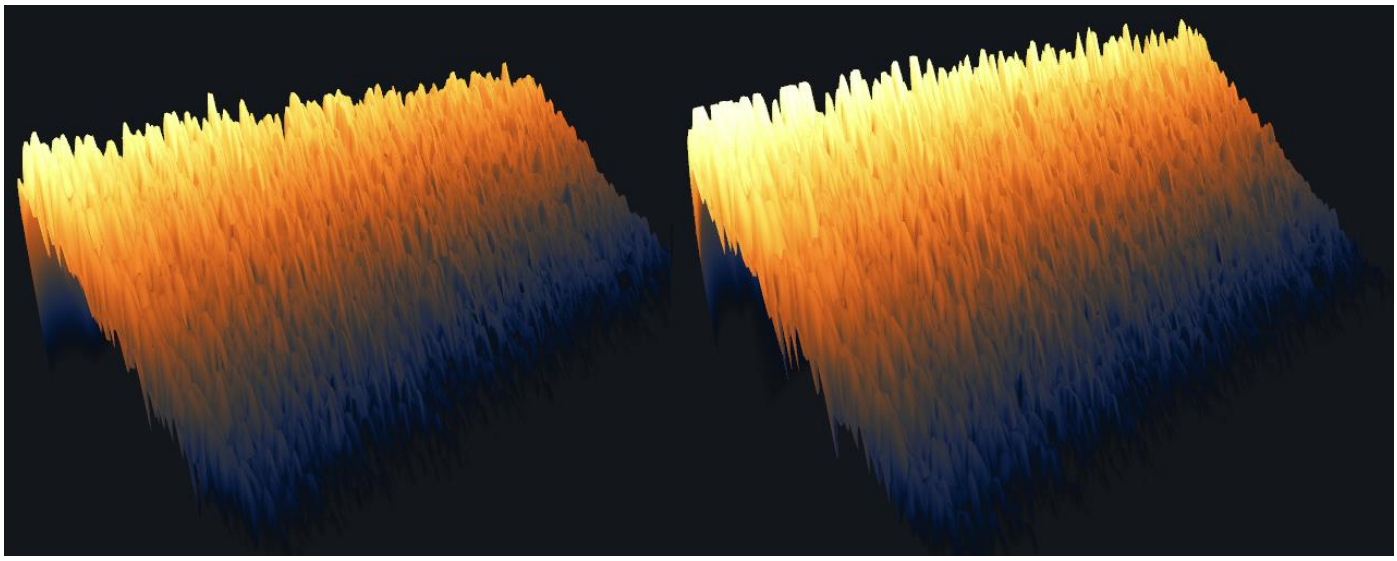
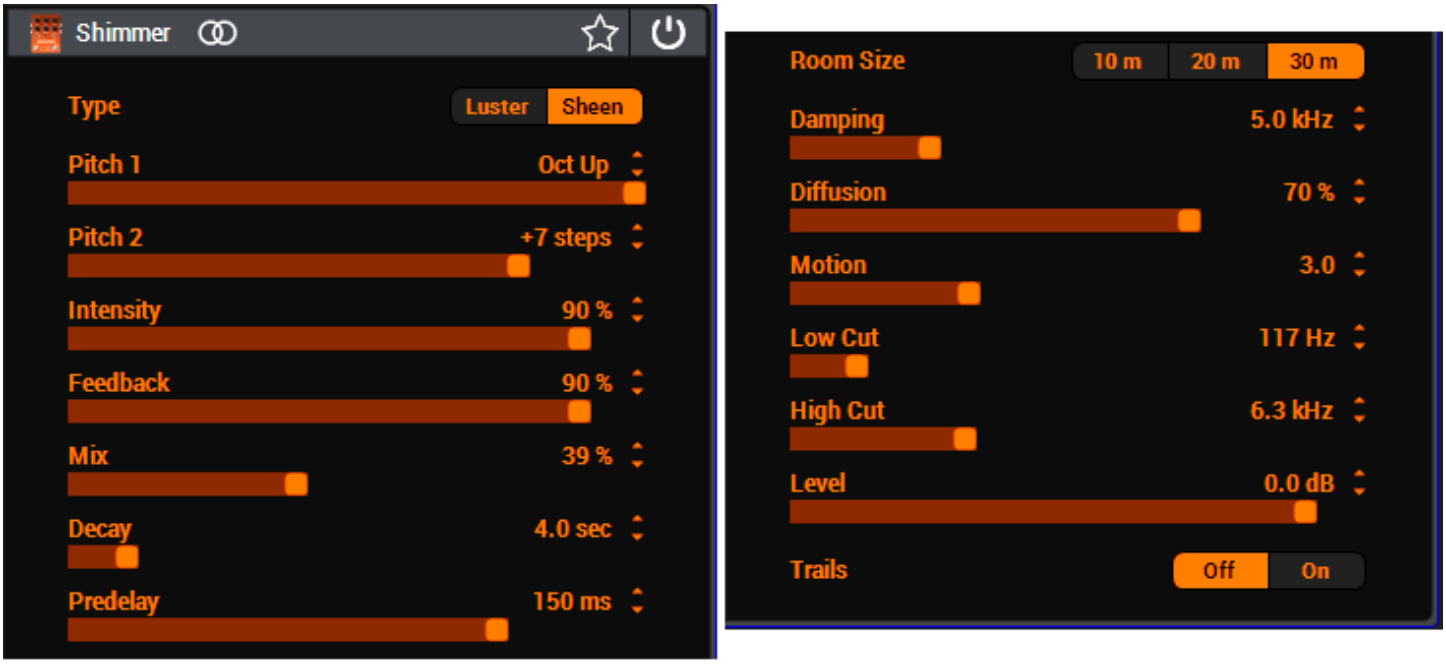


Figure 2.72 No early reflections (left) and early reflections (right).

- **EarlyReflc** stands for “early reflections” (fig. 2.72). In some rooms, the initial reflections from a sound source are discrete echoes that build up and reinforce each other. These fade out into the general reverb tail after a short period of time. The **EarlyReflc** parameter goes from no early reflections (0.0) to significant early reflections (10.0), where more early reflections take longer to decay. In fig. 2.72, note how the reverb on the right has a mass of reflections at the beginning, and their brighter yellow color indicates they’re a higher-level signal than the start of the reverb tail on the left.

Shimmer Reverb



"Shimmer" reverbs inject harmonics from the input signal into the reverb, typically by using real-time pitch shifting. Adding feedback causes the pitch-shifted reverb to return to the input and be pitch-shifted again. This creates a reverb effect whose pitch is constantly stepping up with upward pitch shifts, or down with downward pitch shifts. This interplay of pitch-shifting, feedback, and decay time creates an otherworldly, complex reverb sound that has little to do with traditional room ambience (well, maybe room ambience on Alpha Centauri). This Line 6 original includes two pitch-shifters, as well as two distinct ways of causing the reverb to "bloom."

Mix, **Decay**, **Predelay**, **Low Cut**, **High Cut**, **Level**, and **Trails** work as described in the Reverb Blocks section. **Room Size**, **Damping**, **Diffusion**, and **Motion** are essentially the same controls as for the Dynamic Hall.

The following controls are specific to Shimmer Reverb. Your mileage may vary, but they kept me entertained for hours...

- **Type** chooses between Luster (a tighter, more traditional shimmer effect), or Sheen (expands and extends the bloom). Try them both.
- **Pitch 1** and **Pitch 2** are independent pitch shift effects that shift the input pitch, in cents, from one octave down to one octave up. Although calibrated in cents, semitones are indicated too.
- **Intensity** determines the amount of harmonics injected into the reverb.
- **Feedback** sends some of the audio back to the input, but remember, the audio that's fed back is pitch-shifted. So, the nature and tone of the feedback changes constantly, unless the Pitch settings are 0.0.

Using the Shimmer Reverb

The usual application involves upward pitch-shifting, to give a high-frequency shimmering effect. Traditional reverb tends to lose high frequencies as it decays, especially with damping applied, whereas the Shimmer Reverb's high frequencies (caused by pitch-shifting) extend further in time. This gives a bright, animated sound (fig. 2.73).

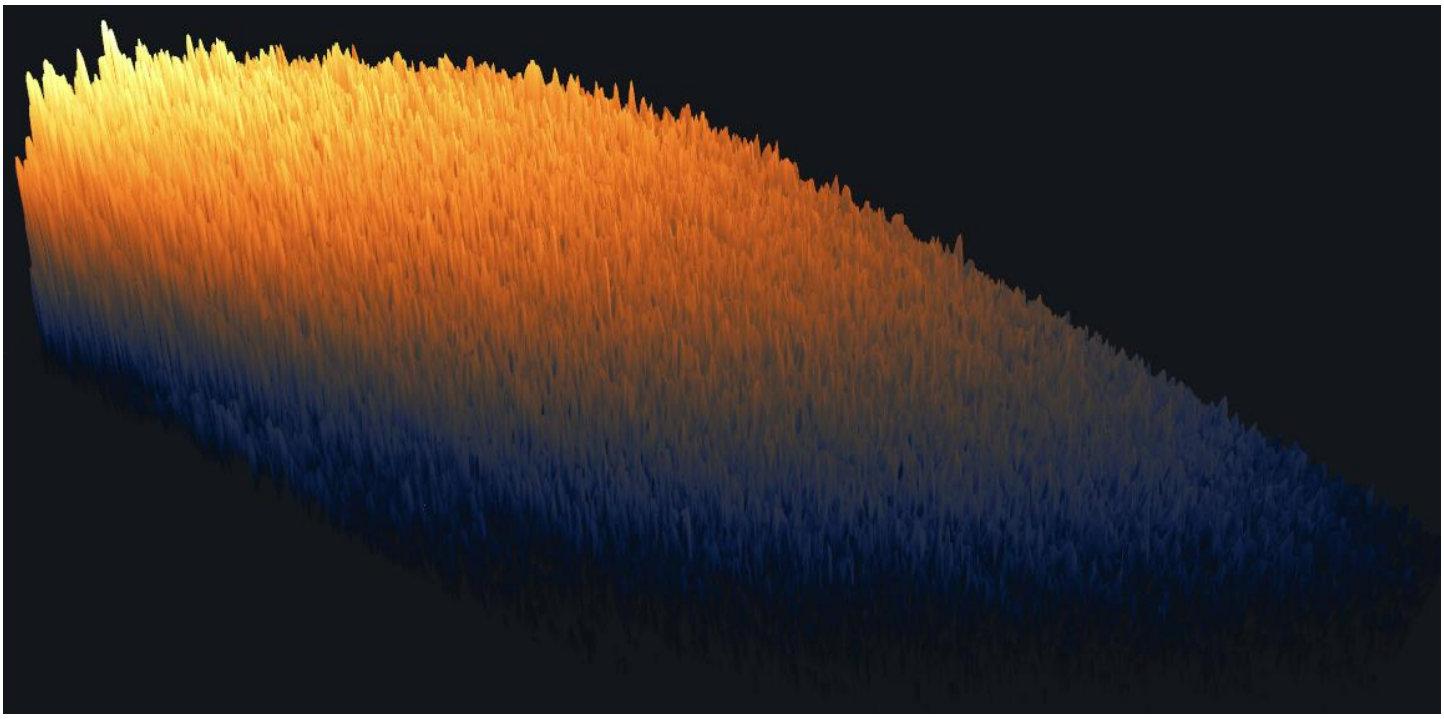


Figure 2.73 Note how applying upward pitch shifting makes the tail's high frequencies last longer than the low frequencies. This image shows Luster mode. Sheen would make the tail visually "fatter."

However, the Shimmer Reverb can also pitch down, or down *and* up simultaneously (remember, there are two pitch-shift parameters). Fig. 2.74 shows an octave pitch up and down, with Luster mode. Note the prominence of the harmonics as the reverb decays, and also, how the reverb maintains a fairly constant level initially, before shimmering into oblivion.

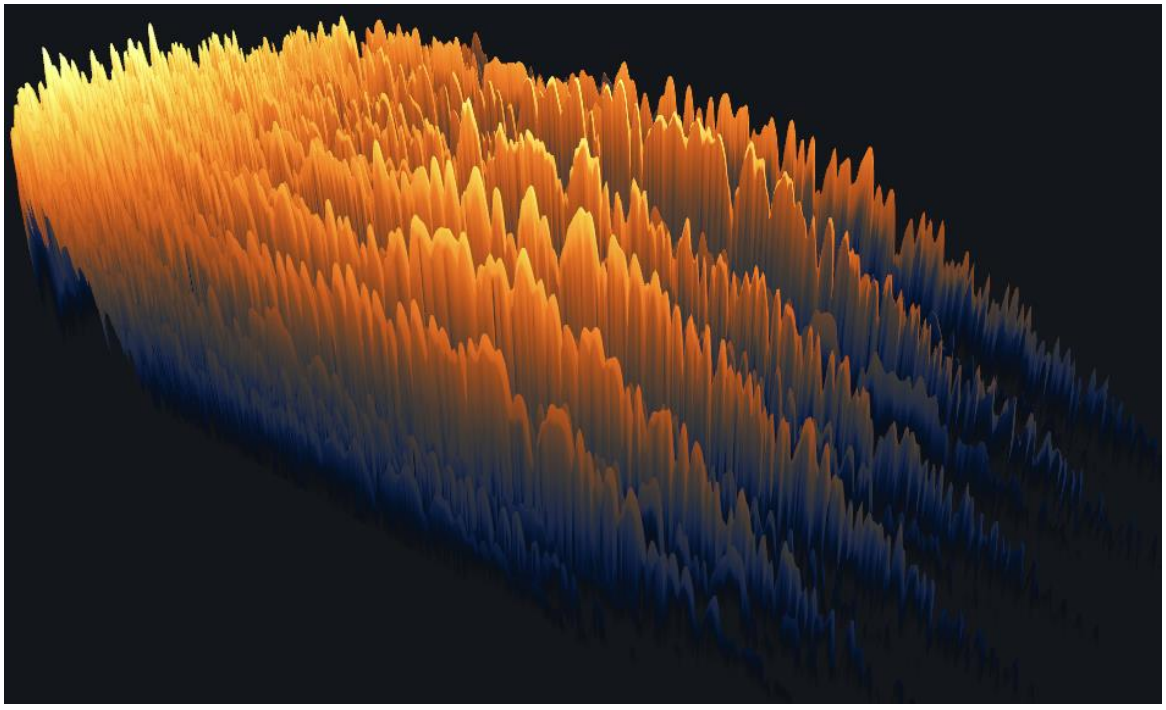
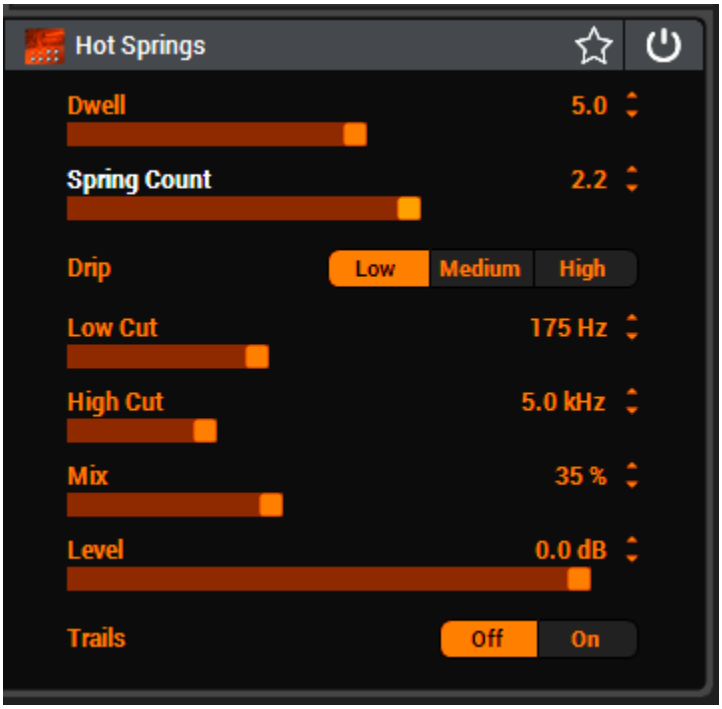


Figure 2.74 Up and down pitch-shifting creates a reverb tail with wild, present harmonics.

Bottom line: This is no normal reverb. Have fun!

Hot Springs

And who doesn't like a traditional spring reverb sound? This block emulates different types of spring reverbs. Note that the Hot Springs reverb doesn't build up over time. Instead, it has an almost instant attack.



The following parameters are spring reverb-specific:

- **Dwell** sets the amount of signal going into the spring tank. Higher values give longer decays (fig. 2.75).

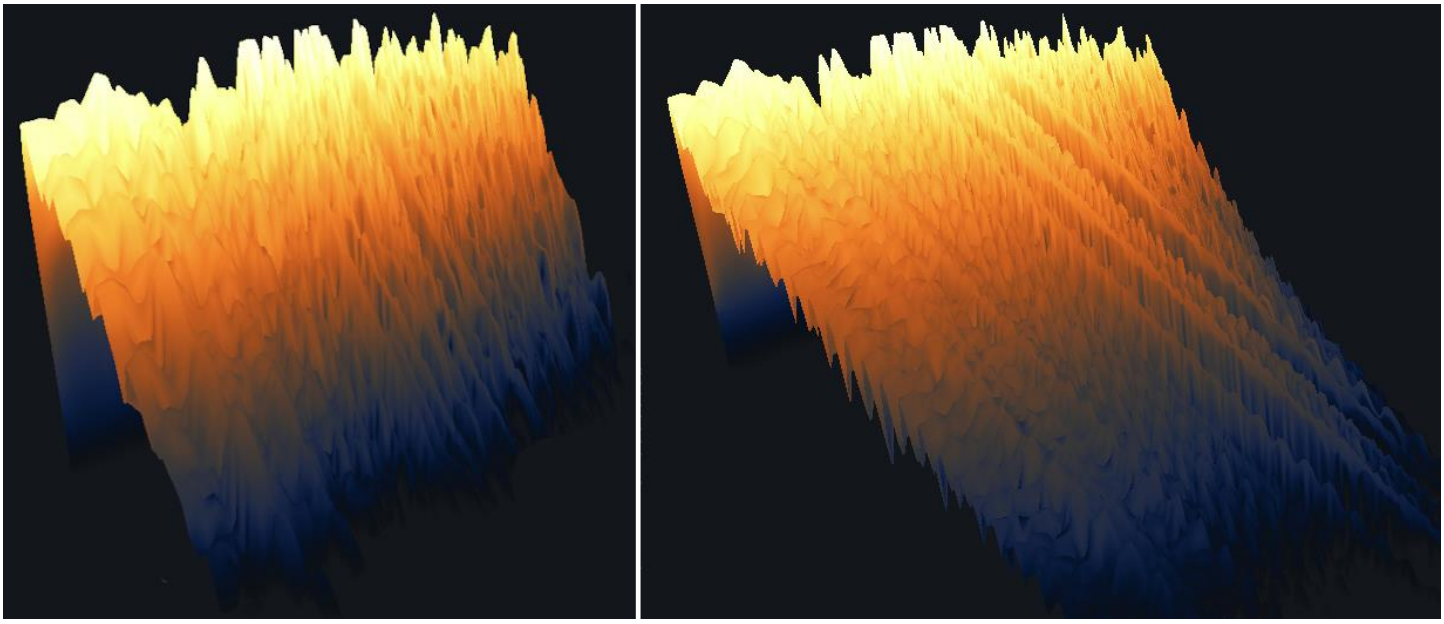


Figure 2.75 Short Dwell (left), and long Dwell (right), with its long, smooth decay.

- **Spring Count** sets the number of springs in the tank from 1 to 3. More springs diffuse the sound more, which becomes somewhat less "spring-like," and more like traditional reverb (fig. 2.76).

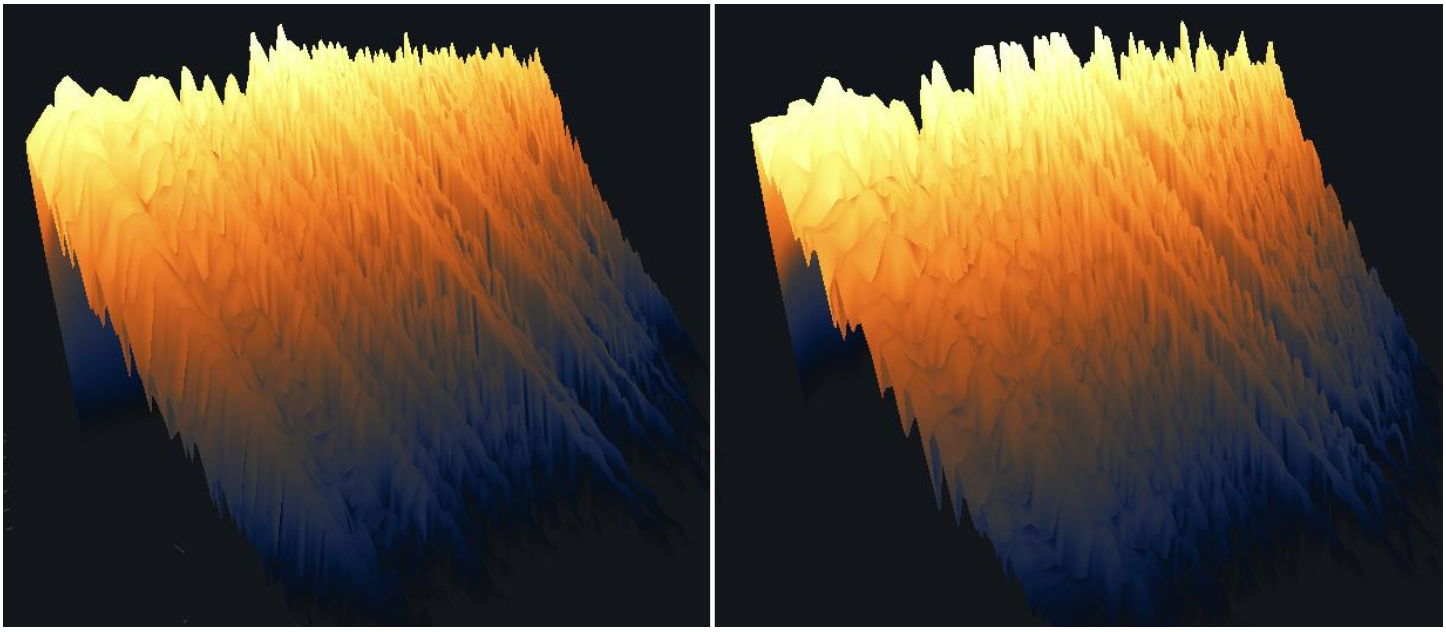


Figure 2.76 One spring (left), and three springs (right). The reverb on the right looks and sounds smoother, particularly at the lower frequencies toward the left side of the reverb's decay.

- **Drip** determines the spring's character. Spring reverbs have a distinctive "sproing" sound. High Drip settings emphasize this more than low Drip settings (fig. 2.77).

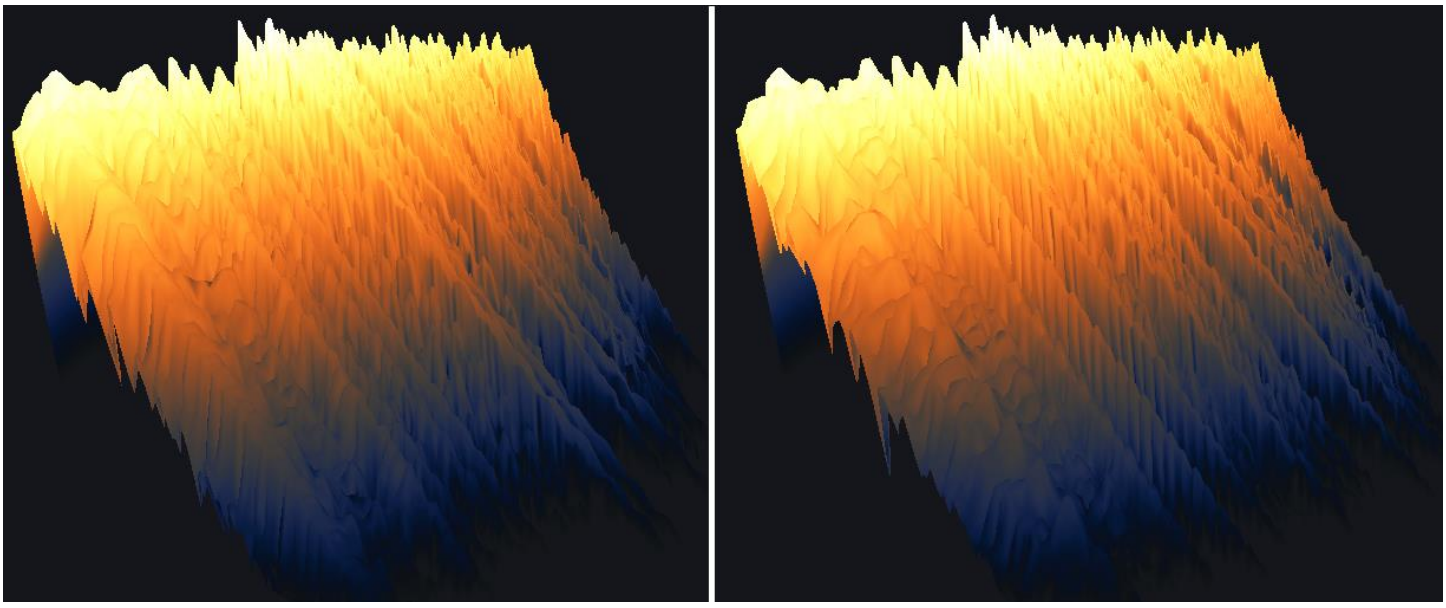
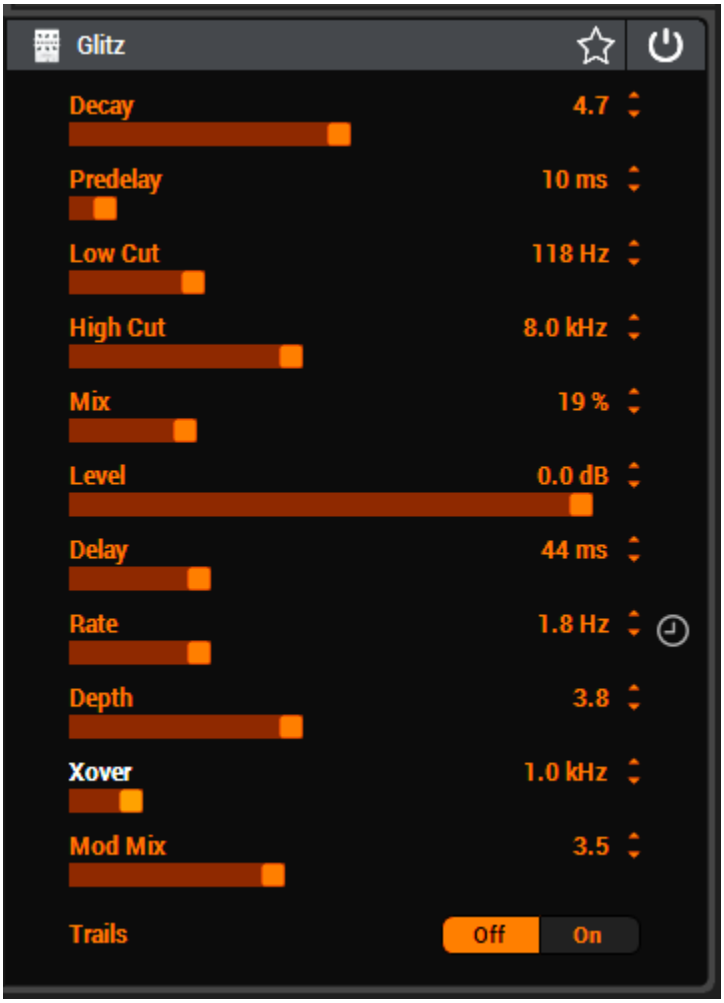


Figure 2.77 Low Drip (left), and high Drip (right). Note the more jagged decay with the high Drip setting, and the discrete "teeth."

Glitz

This uses complex modulation to vary the reverb time, which tends to emphasize different frequencies compared to a typical acoustic reverb. The reverb algorithm also applies a *second* modulation generator to both the Predelay time, and toward the end of the reverb decay. There's no damping parameter, because damping is tied in with the Decay parameter. Glitz's reverb "blooms" (builds up) over more time than the other reverbs.



- **Delay** sets the amount of time before the second modulation source affects the reverb.
- **Rate** determines the speed of the two modulation sources.
- **Depth** controls the amount of modulation that's applied.
- Frequencies above **Xover** are processed with the additional modulation.
- **Mod Mix**, a post-Xover parameter, sets the mix of sounds with and without additional modulation.

As to what that all means, let's look at some Spectrograms before covering how to set the parameter values. In the following images, all the parameters are the same, except for the Xover and Mod Mix parameters. Brighter shades of yellow indicate more level.

Fig. 2.78's Xover frequency is 1 kHz. In the left image, the Mod Mix is at 0.0 (minimum). The amount of reverb and modulation tapers off starting around 1 kHz. The image on the right has Mod Mix at 10.0 (maximum). Now there's a lot more action above 1 kHz, along with a considerable amount of modulation, which causes the waveform's wave-like appearance at high frequencies.

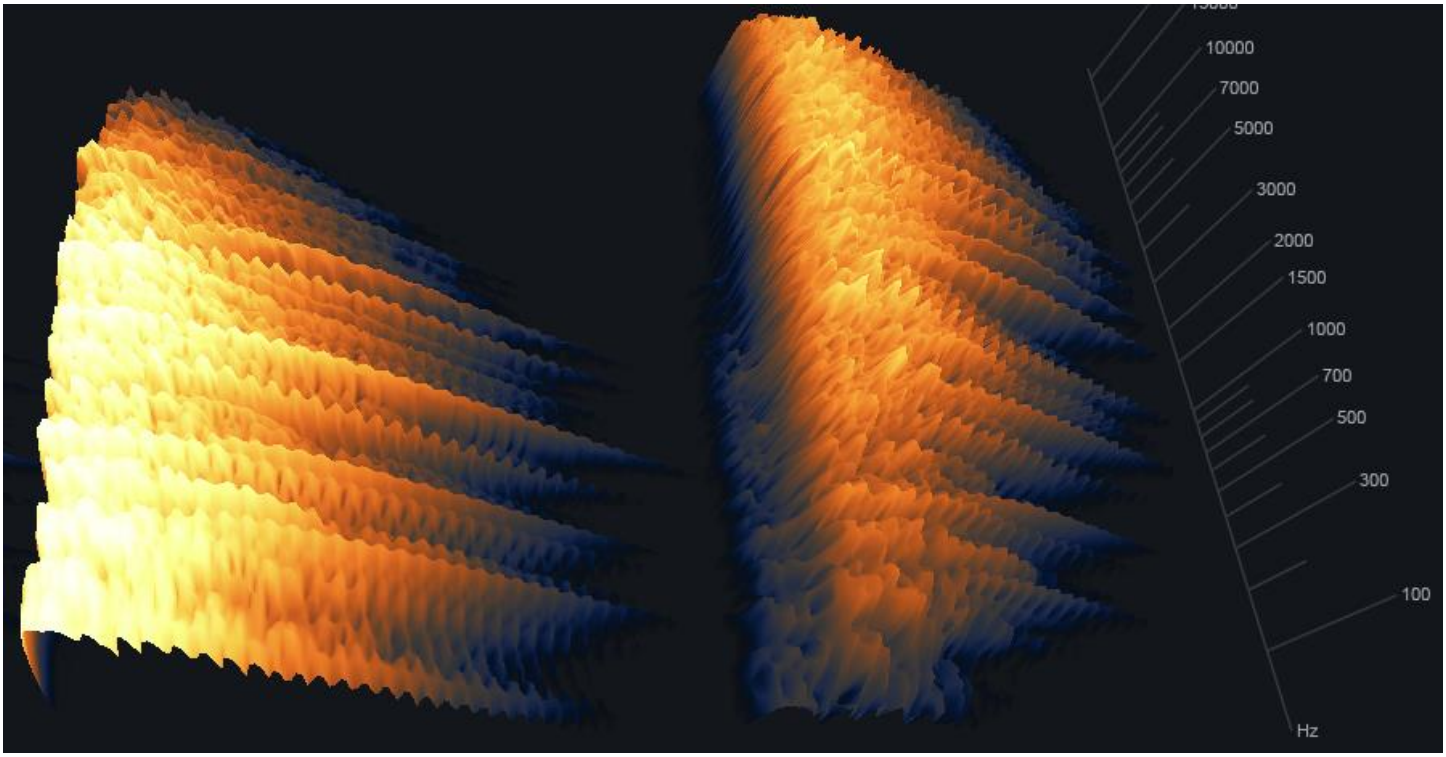


Figure 2.78 Mod Mix is at 0.0 in the left image, and 10.0 in the right. Xover frequency is 1 kHz.

Fig. 2.79 has identical parameters, except the Xover frequency is 200 Hz.

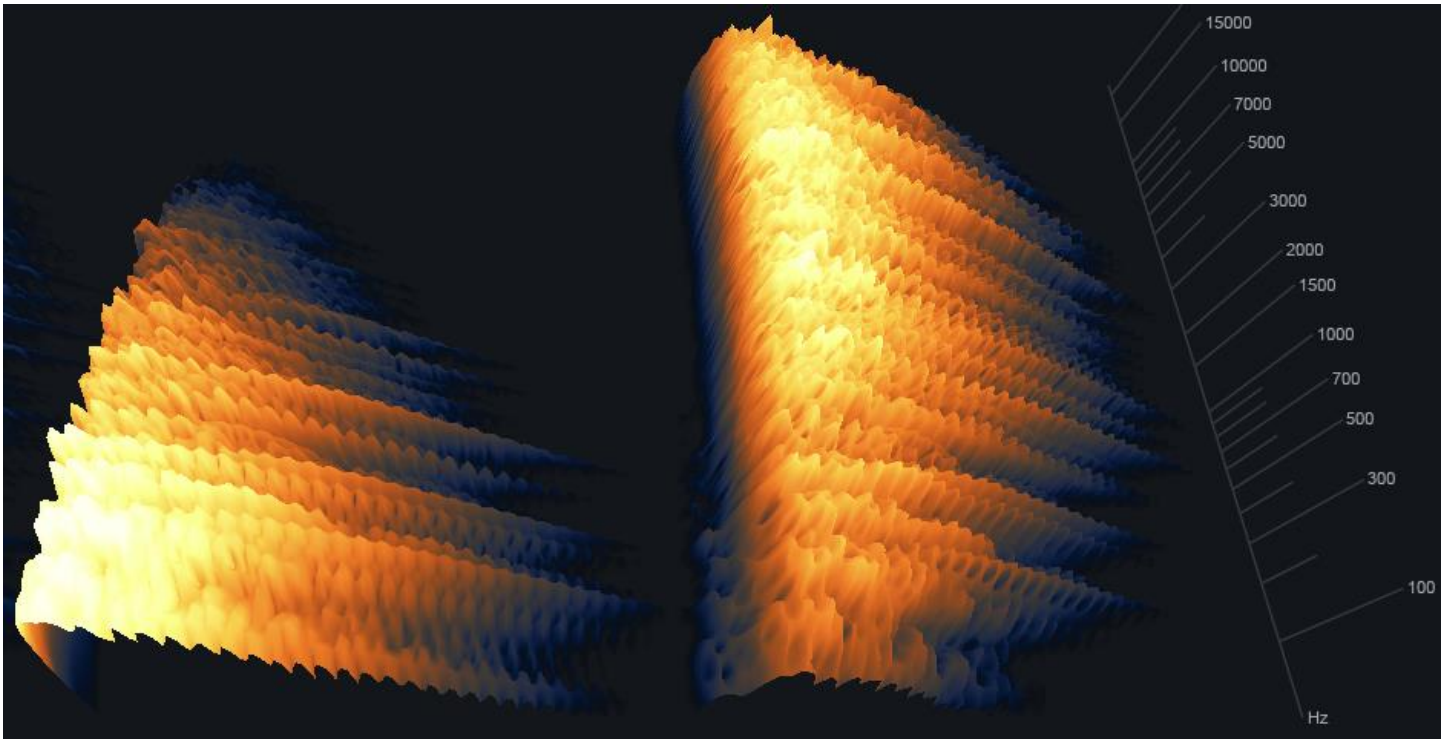


Figure 2.79 Mod Mix is at 0.0 in the left image, and 10.0 in the right. Xover frequency is 200 Hz.

Finally, fig. 2.80 again has the same parameter values, except the Xover frequency is 8 kHz. There's very little low-frequency action, and lots of high-frequency reverb with modulation.

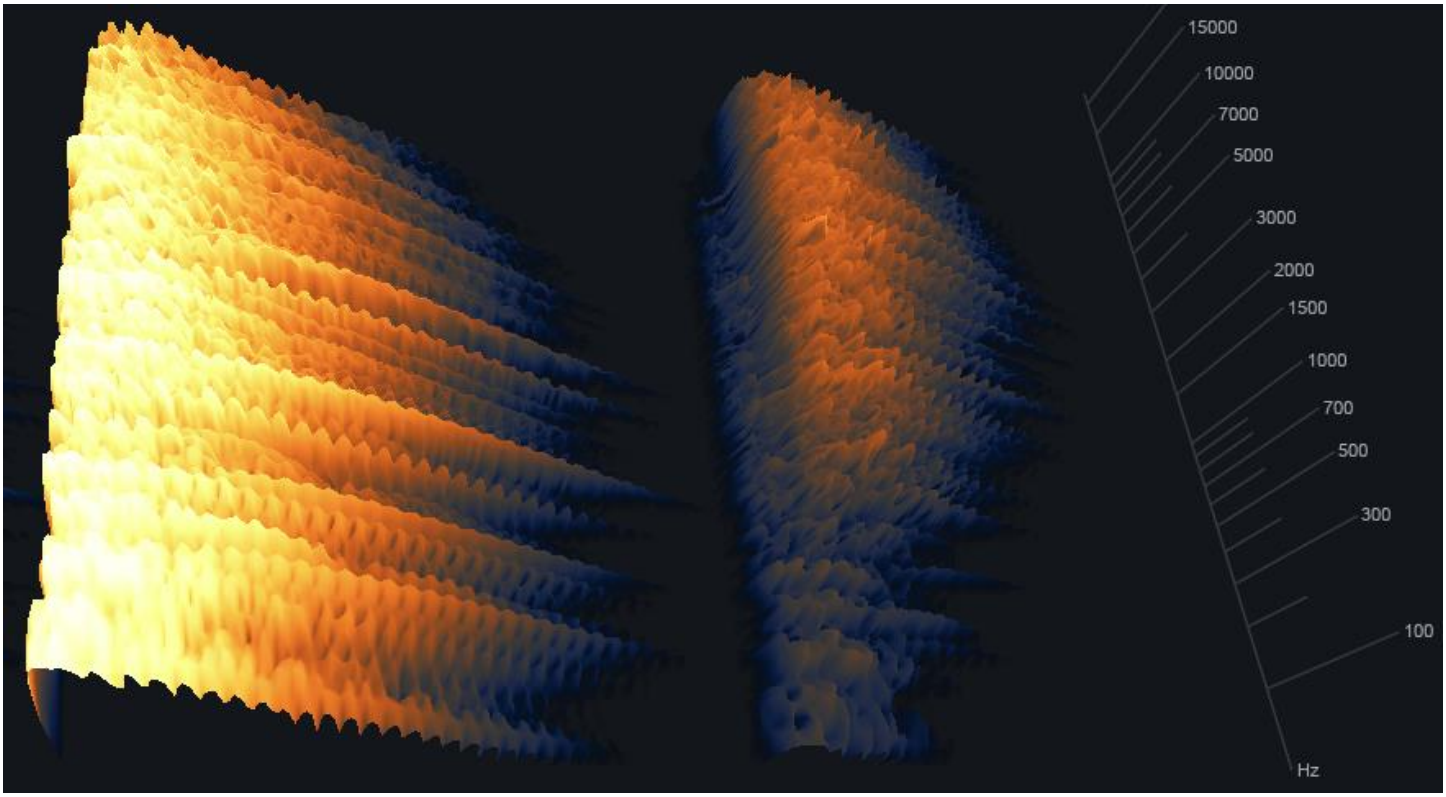


Figure 2.80 Mod Mix is at 0.0 in the left image, and 10.0 in the right. Xover frequency is 8 kHz.

Remember that the emphasis from Mod Mix won't happen until after the Delay time has elapsed. Other interacting parameters include Rate and Depth.

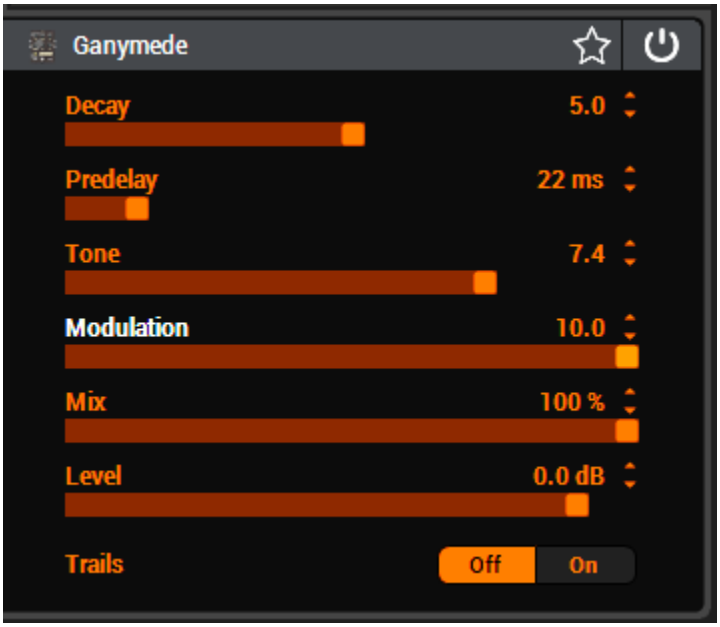
The Bottom Line

Xover Frequency chooses where there will be the greatest emphasis as you turn up Mod Mix. Turning down Mod Mix gives a more even frequency response. For example, for a diaphanous, high-frequency, modulated reverb, you'd set Mod Mix to maximum. Set Xover Frequency fairly high, and then lower the frequency until the reverb covers the desired frequency range.

It's important to realize that all the controls interact. To dive deep into how they affect the sound, feed a short pink noise burst into Glitz, and play with the parameters.

Ganymede

This is a typical stompbox-type reverb, with a somewhat more metallic sound than the Dynamic Hall. It also has a less complex parameter set than other reverbs.



The unique features are:

- **Modulation.** Higher values at longer Decay times give a smoother reverb tail, with less periodicity.
- **Tone** alters the frequency response. Higher values give more highs, and less damping (fig. 2.81).

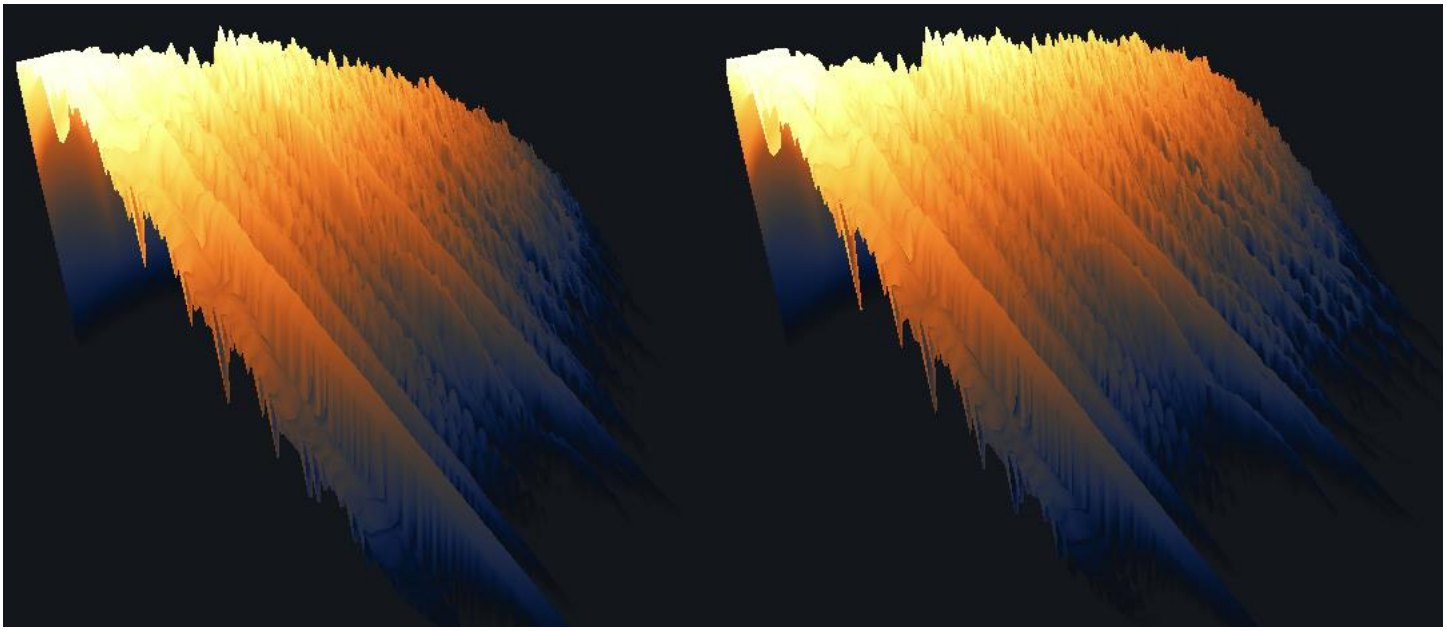
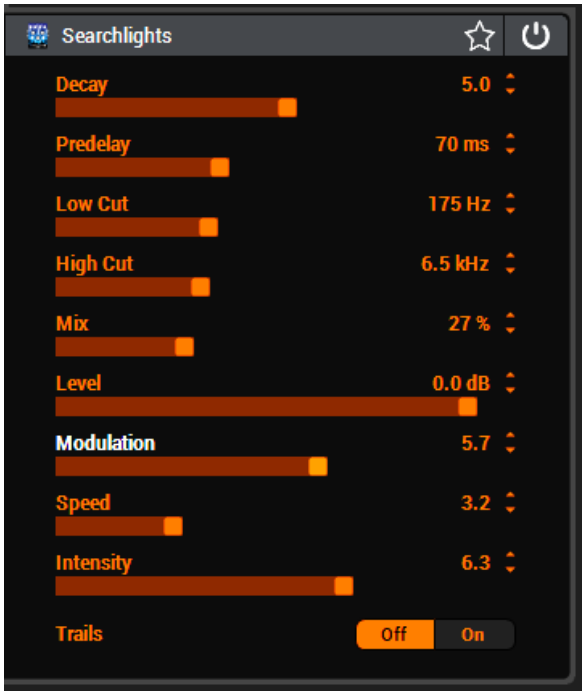


Figure 2.81 Low Tone setting (left). Note the faster and less prominent high-frequency decay, and stronger bass component, compared to the high Tone setting (right).

Searchlights

Turning up the Modulation amount and Intensity settings adds a phaser-like sound quality. The parameters are standard, aside from three unique controls:

- **Modulation** determines the “shape” of the modulation that affects the reverb.
- **Speed** sets the modulation rate.
- **Intensity** determines the extent to which the modulation affects the reverb.



This is a “special effects” reverb—note the extreme reverb decay variations in fig. 2.82.

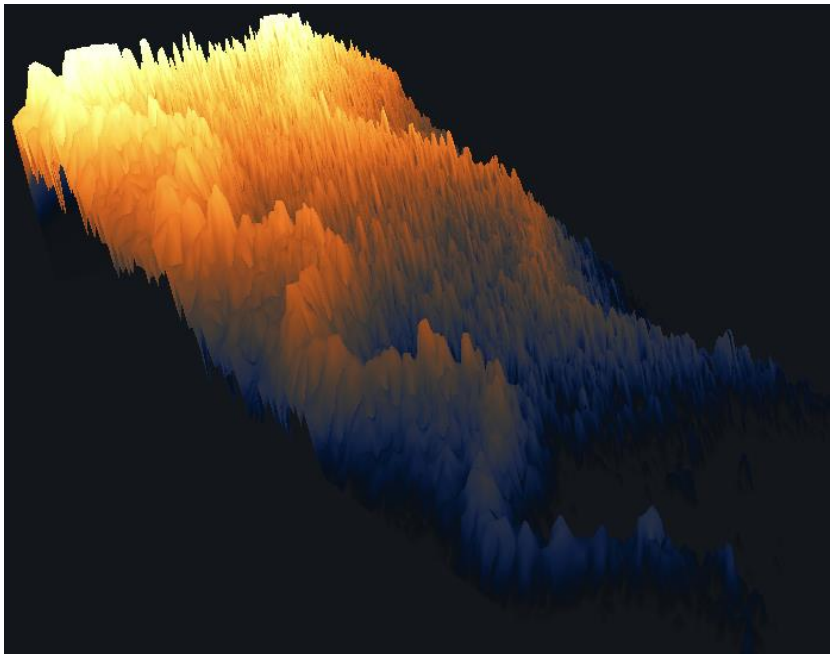
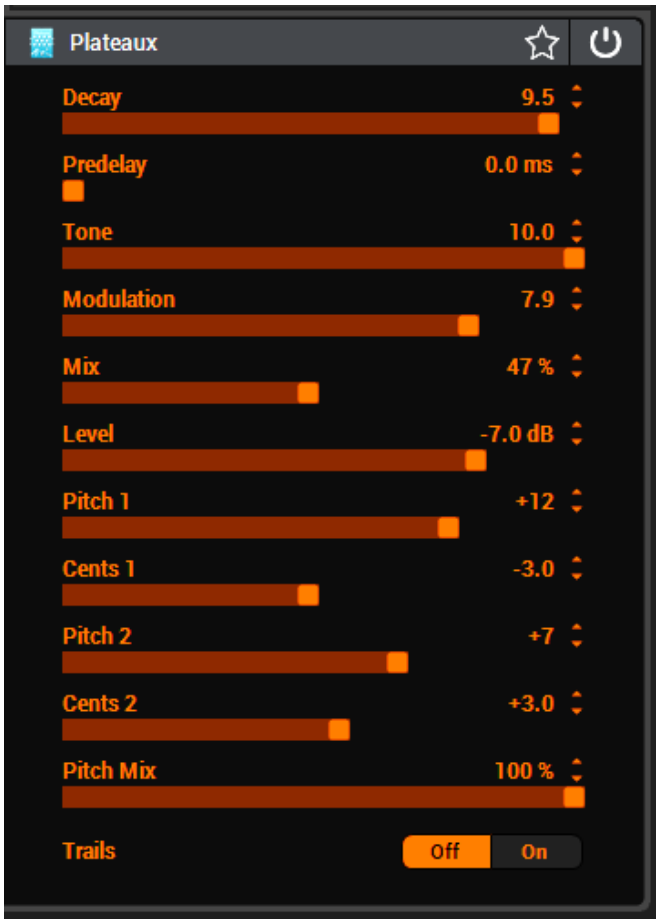


Figure 2.82 This spectrogram shows the results of fairly high modulation and intensity settings. The reverb tail frequency response and levels vary dramatically over time.

Plateaux

“Shimmer” reverbs pitch-shift the reverb, typically through feedback to the input, with the feedback offset by a few cents. Offsetting the feedback pitch creates an ongoing pitch change as the pitch-changed feedback recirculates. The resulting sound incorporates elements of reverb, pitch-shifting, and chorusing. The main application is with ambient musical styles.



The controls are different from traditional reverb. It can take a little effort to dial in the desired sound.

- **Modulation** sounds like vibrato at a relatively fast rate.
- **Pitch 1** (adjusted in semitones) imparts a sense of pitch to the reverb. **Pitch 2** is independent but works the same way.
- **Cents 1** fine-tunes the pitch. The pitch bends further as the reverb feeds back. Smaller values give a chorusing-like effect, while higher values sound detuned. **Cents 2** is independent but works similarly.
- **Pitch Mix** sets the amount of pitch-shifted sound that’s included in the overall reverb effect.

Tips

- A Decay setting of 10.0 isn’t infinite, but it’s close.
- Changing Modulation while the decay is still occurring gives some wild pitch-bending effects. Consider assigning this parameter to an Expression Pedal.
- The Shimmer effect is most prominent with long decay times.
- For standard reverb sounds, set Pitch Mix to 0% and modulation to 0.0.
- Plateaux has the same basic reverb engine as Ganymede.

Double Tank

With Modulation at 0%, this is a traditional reverb with a quick attack (it doesn't sound like there are many early reflections) and conservative damping. But experimenting with the sweet spot for modulation can give a smooth, "linear" kind of decay. The crucial controls are:

- **Rate** sets the modulation speed. Higher percentages tend to give smoother reverb tails.
- **Modulation** determines how much the Rate influences the decay.

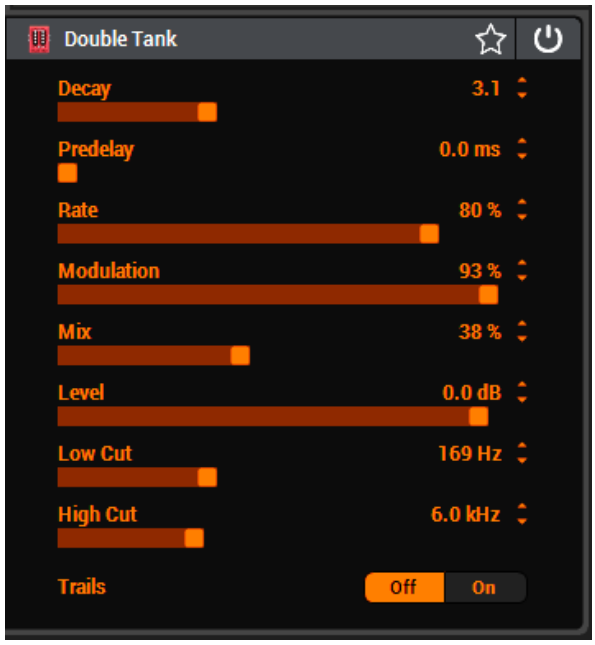


Fig. 2.83 shows Double Tank reverb settings without modulation (left) and smoothed by modulation (right), using the settings shown above.

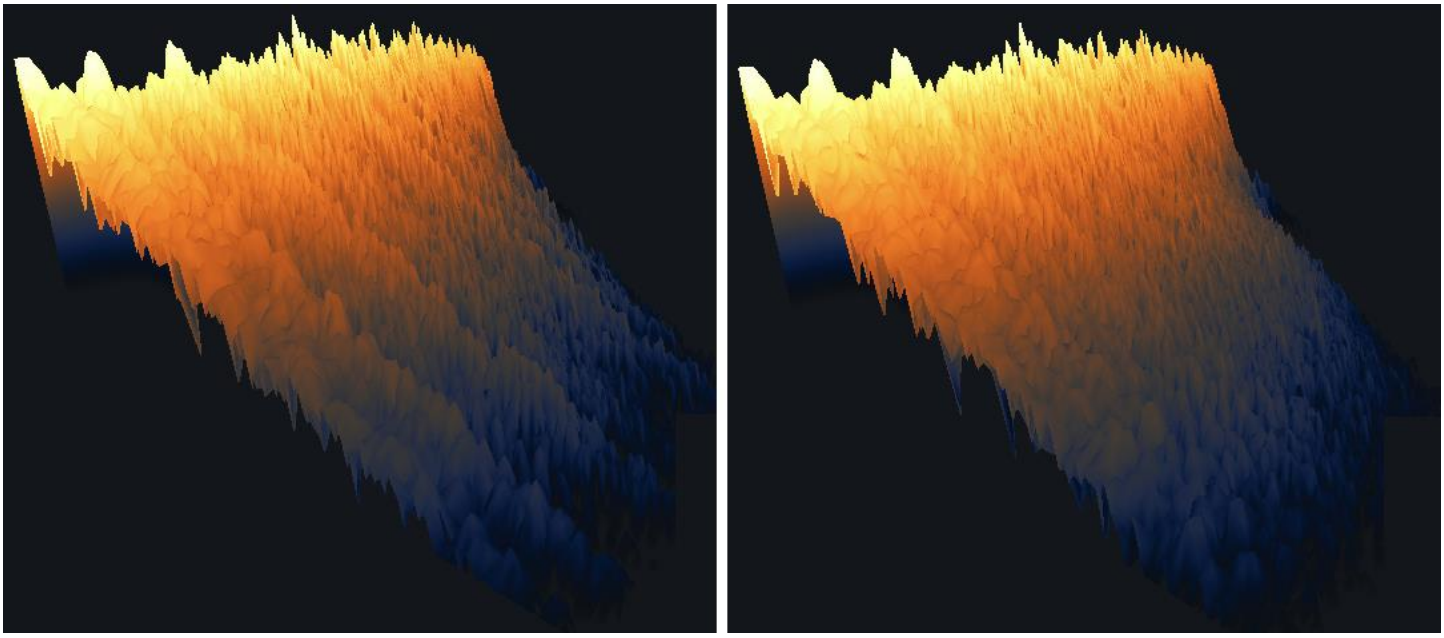
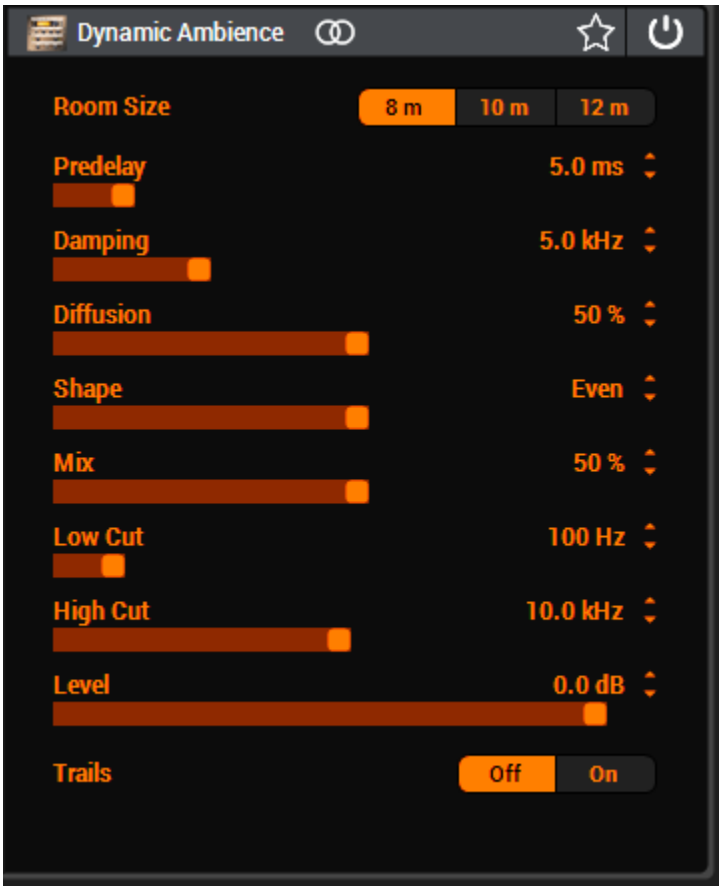


Figure 2.83 Adding modulation smooths out the reverb tail on the right.

Dynamic Ambience

This specialty reverb emulates the sound of playing in a small acoustic space, compared to the larger room size the Dynamic Room effect provides. Ambient reverb consists of myriad short reflections that decay very quickly, not a long reverb "tail." Ambient effects are ideal for adding a more natural feel to dry sounds. Although some reverbs can create a similar effect, they require more processing power and aren't optimized for this application. The Dynamic Ambience processor also avoids the potential phase problems involved in using short stereo delays to give ambiance.



In addition to the parameters covered in the Reverb Blocks section (**Predelay**, **Mix**, **Low Cut**, **High Cut**, **Level**, and **Trails**), Dynamic Ambience includes:

- **Room Size** creates three different rooms (calibrated in meters). **8m** gives the most close-in feel. **12m** opens up the room the most.
- **Damping** determines how the reflections' high frequencies decay. Higher frequencies give a brighter decay, as if the virtual room had hard surfaces. Lower frequencies give a darker decay, as associated with a room that has curtains, people, rugs, or other materials that dampen the sound.
- **Diffusion** sets the spacing between when echoes occur. More diffusion packs the echoes closer together. This smooths the sound of the reflections. Less diffusion creates reflections that are more like a discrete series of echoes than a single, cohesive sound.
- **Shape** sets the balance of the very first early reflections, compared to the late reflections that occur as the original reflections bounce off the room surfaces (fig. 2.84). This parameter's **Even** setting produces an equal balance of early and late reflections.

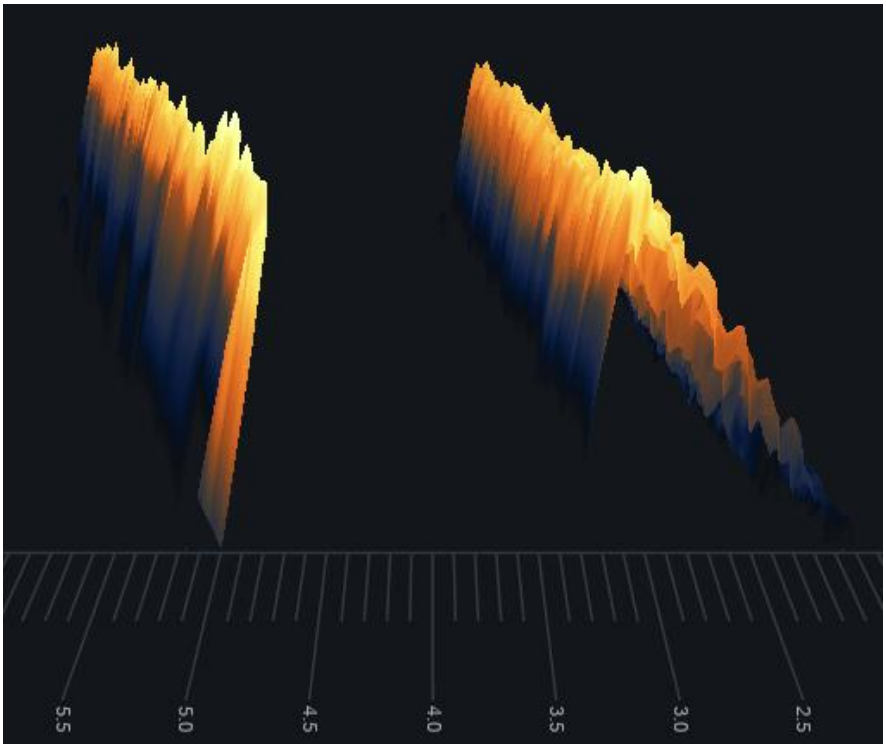


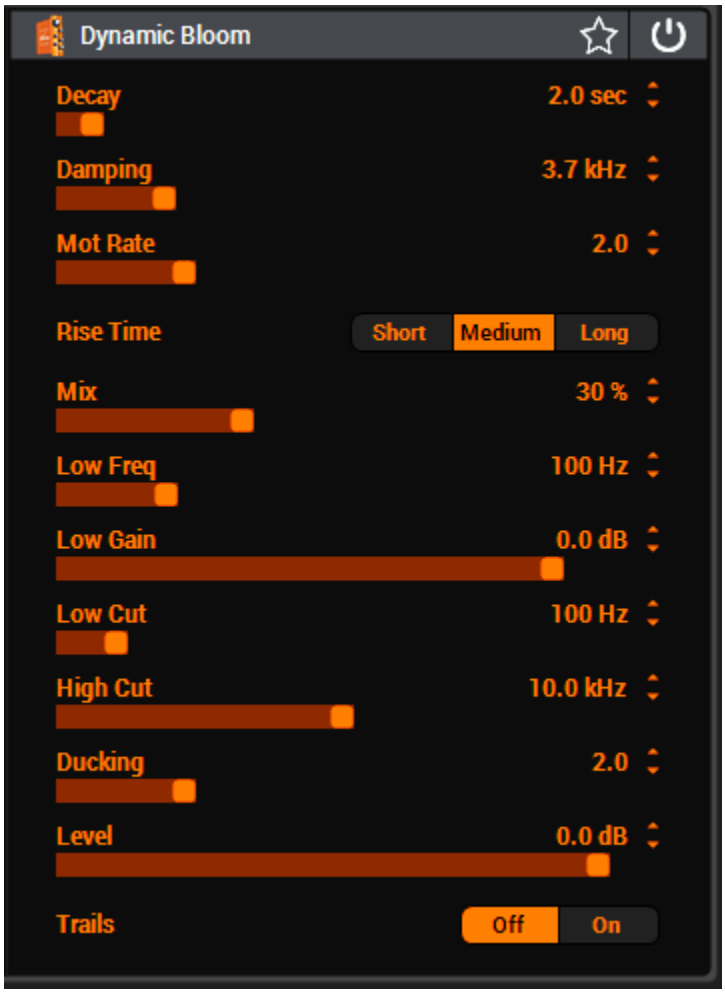
Figure 2.84 The image on the left shows the initial reflections, with Shape set to **Early 100** (i.e., 100% early reflections). The image on the right shows Shape set to **Late 100** (i.e., 100% late reflections). Both images are on the same baseline, but in the right image, note that there are almost no early reflections.

Tips

- For the smoothest sound, set **Damping** between 5 and 8 kHz, **Diffusion** to 100%, and **Shape** to Even.
- **Shape** interacts with Room Size. Choosing more late reflections with Shape increases the apparent room size. The biggest ambient effect occurs with Room Size set to 12m, and Shape set to Late 100.
- Those who use Helix Native as a studio plug-in can take advantage of Dynamic Ambience to bring dry drum loops and analog drum sounds to life. Giving them an acoustic space greatly increases realism.
- Dynamic Ambience is an *excellent* choice to put at the end of a preset when you want to open up the sound, but conventional reverbs require too much processing power.

Dynamic Bloom

A “bloom” reverb has an attack time that stays out of the way of note attacks. The reverb builds up over less than a second before decaying like a standard reverb. I find this effect so useful that in the past, I’ve created bloom reverb setups by using two reverbs, phase changes, and different decay times. Now all I need is the Dynamic Bloom reverb. It can provide great effects for far more than just guitar. Try Dynamic Bloom with vocals and melodic percussion.



In addition to the parameters covered in the Reverb Blocks section (**Mix**, **Low Cut**, **High Cut**, **Level**, and **Trails**), Dynamic Bloom includes:

- **Decay** works as described in the Reverb Blocks section. Its shortest decay time is 0.1 seconds. **Tip:** If the rise time is longer than the decay, you won’t hear much reverb because the reverb will have faded out before the rise can bring the reverb up to its full level.
- **Damping** determines how the reverb’s high frequencies decay. Higher damping frequencies give a brighter decay, as if the virtual room had hard surfaces. Lower damping frequencies give a darker decay, as associated with a room that has curtains, people, rugs, or other materials that dampen the sound.
- **Mot(ion) Rate** is somewhat like a subtle chorus. Higher speeds smooth the reverb more, which is most noticeable on tails with higher-frequency damping settings and long decays.
- **Rise Time** (fig. 2.85 and 2.86) sets the time for the reverb to bloom to its full amount.

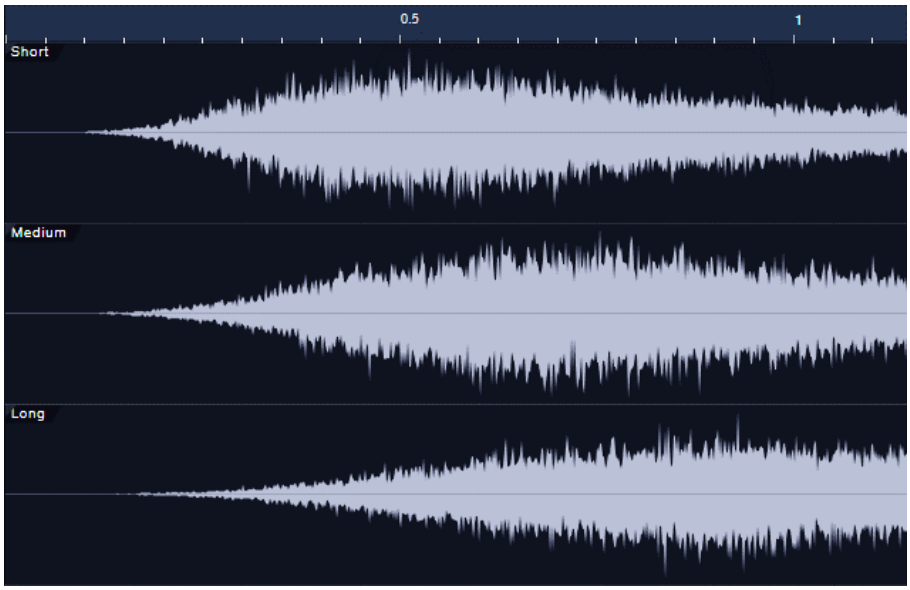


Figure 2.85 The scale on the top is in seconds. The Short setting reaches maximum reverb in about 400 ms. Medium takes about 700 ms, and Long is about 900 ms.

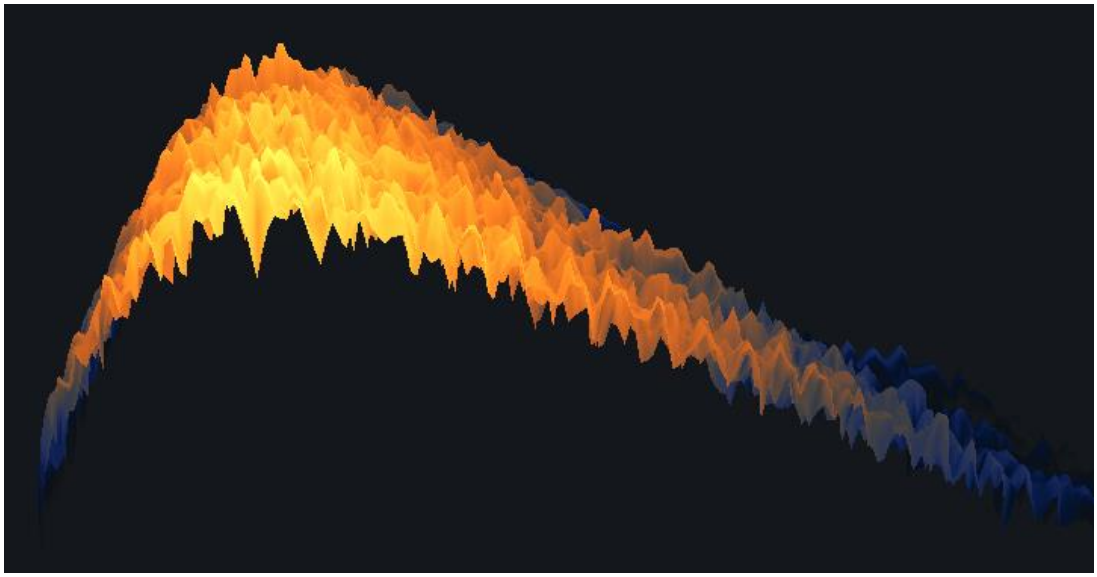
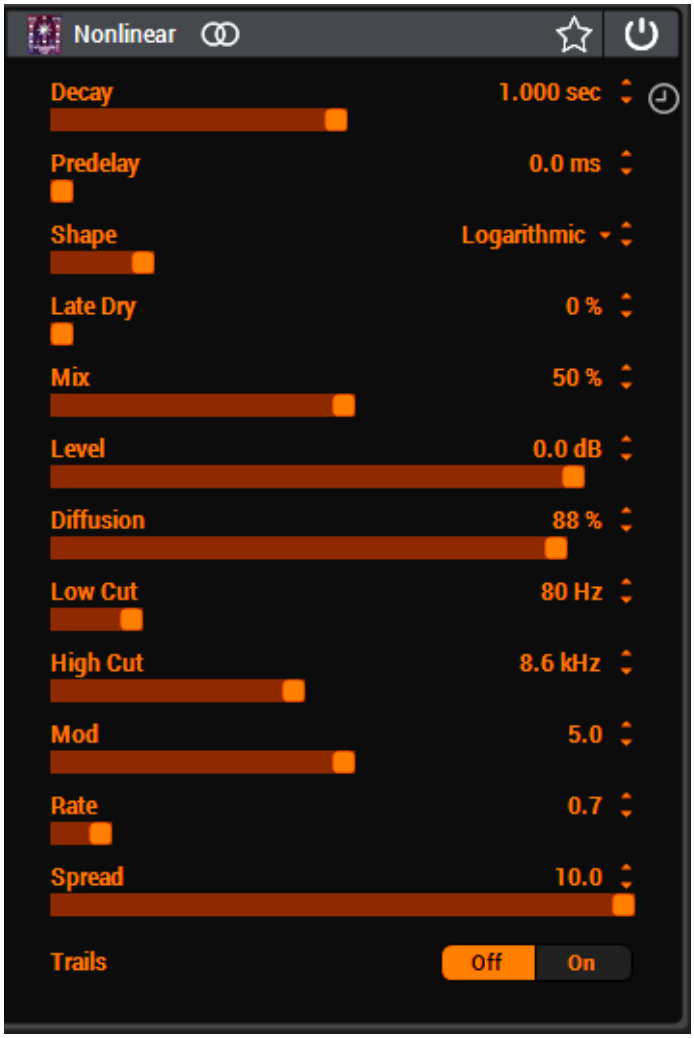


Figure 2.86 Unlike previous reverb spectrum images, this shows the side view for the reverb's decay. With a Long Rise Time setting, you can see how the reverb "blooms" to its full level.

- **Low Freq** determines a crossover frequency. Below this frequency, Low Gain increases the reverb's decay time in this frequency range.
- **Low Gain** sets the reverb time for frequencies below the Low Freq value. The bass frequencies decay faster than treble frequencies for values below 0.0 dB. This gives a "tighter" reverb sound with less potential for mud. With values above 0.0 dB, the bass frequencies decay more slowly than the treble frequencies. This can give a bigger sound and the sense that there's more damping.
- **Ducking** lowers the gain of an existing reverb tail when a new reverb tail appears. With low ducking settings, successive tails continue to their full decay. With high ducking settings, a new tail reduces the level of tails that are still decaying. This prevents overlapping tails from stepping on each other. **Tip:** The maximum setting of 10.0 produces an effect like the "pumping" used in EDM. This occurs because ducking mutes the existing reverb tail(s) before the new tail appears.

Nonlinear

"Nonlinear" means that the reverb doesn't decay in a normal linear fashion. Instead, the reverb can go in reverse, or even conform to various shapes. Like many other Helix effects, this isn't just for guitar. It's a novel tool for sound design.



In addition to the parameters covered in the Reverb Blocks section (**Decay, Mix**, **Low Cut**, **High Cut**, **Level**, and **Trails**), Dynamic Bloom includes:

- **Shape** chooses the shape of the reverb's decay (fig. 2.87). You have a choice of eight shapes:
    - *Linear:* Reverse reverb ramps up linearly to the maximum reverb level, then stops abruptly at the end of the decay's duration.
    - *Log:* Reverse reverb ramps up slowly at first, then more quickly toward the decay's end.
    - *Inverse Log:* Reverse reverb ramps up quickly to the maximum reverb level, then slows down.
    - *Gauss:* Ramps up and then down in a gaussian curve shape.
    - *Inverse Gauss:* Ramps down and then up in a gaussian curve shape.
    - *Triangle:* Ramps up and then down in a triangle shape.
    - *Inverse Triangle:* Ramps down and then up in a triangle shape.

- *Full:* There's no ramping. The reverb starts with reverb at maximum, and stays at maximum until it stops abruptly at the end of the decay's duration.

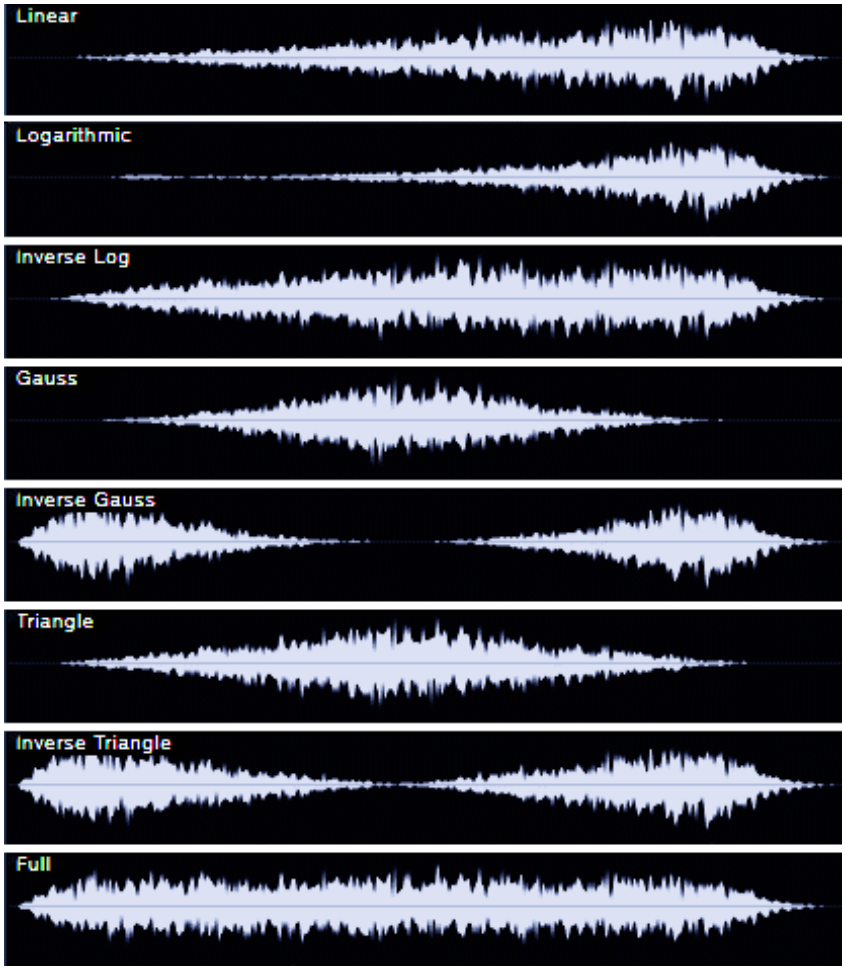


Figure 2.87 Reverb waveforms with names for the various shapes.

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- **Late Dry** mixes in a bit of the original signal as the very last tap. This effect is most audible when playing simple lines with gaussian and triangle shapes.
- **Diffusion** sets the spacing between when echoes occur. More diffusion packs the echoes closer together. This smooths the sound of the reflections. Less diffusion creates reflections that are more like a discrete series of echoes than a single, cohesive sound. For example, low diffusion is common with vocals because vocals are a sustained sound. High diffusion is common with percussion.
- **Mod** applies a flanging-like effect to the reverb. Higher settings increase the effect's depth.
- **Rate** controls the speed of the modulation that's applied to the reverb.
- **Spread.** With the stereo version, this determines the reverb's stereo width. Higher settings widen the stereo image.

Tips

- With Decay set to note values, playing a note/chord 2 or 4 beats before a song transition, with Decay set to "1/2" or "1/1" respectively, causes the reverb to stop right on the downbeat.
- This reverb requires a lot of processing power. But if you don't need to use a lot of other effects, try running two Nonlinear reverbs in parallel, set to different shapes. Pan one reverb to the right and one to the left. The sound is pretty…twisted.

Pitch/Synth Blocks

This section covers effects other than those based on the polyphonic processing features added in Helix version 3.0. The poly feature is so important that it has its own chapter, which goes deep into applications.

Pitch-Shifted Sound Quality

Shifting pitch is a technological challenge. Obtaining high sound quality, accurate pitch detection, *and* low latency is daunting. Any pitch-shifting algorithm has to make tradeoffs. For example, it's possible to obtain high sound quality—if you're willing to wait 150 milliseconds or so to hear the note you're playing.

Real-time pitch shifting for live performance is *far* more challenging than the non-real-time processing done in DAWs. Here are some of the tradeoffs:

- More accurate pitch detection and tracking require higher latency. To transpose note pitches accurately, the system needs more time to analyze the note frequencies.
- Sound quality decreases as the transposition amount increases.
- Live performance requires low latency, which forces a tradeoff for lower sound quality.
- High-quality, low-latency, real-time pitch transposition requires a lot of processing power.
- *You* need to make some tradeoffs as well—like playing cleanly (especially with attacks).

Helix's Polyphonic transposition is amazingly good, but that's because it uses so much processing power. This limits the number of potential effects in an effects chain. The pitch-shifting effects described in this section use less processing power, so they're suitable for more complex effects chains.

Optimizing Sound Quality

With pitch-shifters designed for single notes, playing more than one note at a time can give a hot mess of a sound. Playing cleanly improves the sound quality. By definition, pitch shifters that generate harmonies create more than one note at a time. For these, effects chain placement is crucial:

- If an effects chain includes effects that generate distortion (including amps), it's usually best to insert the pitch shifter *after* distortion. However, if the sound quality is acceptable when playing chords without a pitch shifter, inserting the pitch shifter before distortion may work.
- Inserting the pitch shifter between an amp and cab often sounds better than inserting after an Amp+Cab. This is true for all the non-polyphonic pitch processors.
- With a clean effects chain, most of the time it's best to insert the pitch shifter at the chain's beginning, because then it gets the cleanest possible signal.

Finally, remember that sound quality isn't everything. If you use pitch shifting to give a dramatic dive bomb effect in a sea of distortion, the audience won't care if the audio isn't pristine.

Pitch Wham

The Pitch Wham effect's sound quality is best with single notes, and can do deep bends when you don't have a vibrato tailpiece. Pitch Wham can also bend upward over a wide range, which vibrato tailpieces can't do.

Pitch Wham works well for dive-bomb effects, where you can usually get away with chords (instead of single notes) and lower-fidelity audio, because the effect itself is compelling and momentary. The screenshot shows Pitch Wham set up for a deep, two-octave dive-bomb effect.

- Pitch Wham defaults to Expression Pedal 1, and **Position** represents the Expression Pedal position, from 0% (all the way back) to 100% (all the way forward). You can modify the minimum and maximum position settings for the pedal's full travel in the Position parameter's Controller Assign menu.
- **Heel Pitch** sets the pitch when the pedal is fully back. In the dive-bomb example, 0 means that there's no pitch shift when the pedal is back.
- **Toe Pitch** sets the pitch when the pedal is fully forward. -24 semitones drop the pitch two octaves.
- **Mix** determines the level of dry and pitch-shifted sound.

Tip: I set the Heel Pitch to 0 transposition, and Toe Pitch for the desired amount of shift (up or down). Being able to shift both up *and* down means that returning to pitch will happen at some point along the Expression Pedal's travel. That makes it very difficult to return exactly to pitch. With Heel Pitch at 0, pulling back the pedal all the way guarantees returning to the original pitch. Similarly, you could set to Toe Pitch to 0 if you prefer pushing the Expression Pedal down (instead of pulling back) all the way to return to the original pitch.

Other Applications

Single-note pitch bends and dive-bombs are the main applications. Here are three others:

- **Auto-flanging.** Set Position = 51%, Heel Pitch = +1, Toe Pitch = -1, and Mix = 50%. Because the Pitch Wham's delay changes over time as it works to "splice" audio together to create pitch changes, there's an automatic flanging effect. Position settings up to about 58% give different flanging effects. This same principle can also create a Through-Zero Auto Flanger,
- **Pseudo-steel guitar bends.** To glide up to notes or chords, set position to 100%, Heel Pitch to -1 (or -2), Toe Pitch to 0, and Mix to 100%. I normally play with the pedal in the toe position, then pull back and slide up to a chord or note. You might prefer the reverse pedal motion, where the Toe Pitch is -1 or -2, and the Heel Pitch is 0, so that pulling back on the pedal bends up to pitch.

- **Low tunings** (fig. 2.88). Because sound quality is best with small amounts of transposition, tuning down 1 or 2 semitones can still give acceptable sound quality when your insert Pitch Wham between a distorted amp and cab (this sometimes works before the amp as well—try both to determine which sounds better). Transposing up one or two semitones can also work. The amp's distortion masks the Pitch Wham's artifacts. A downside is that the pitch won't be the same as your strings.

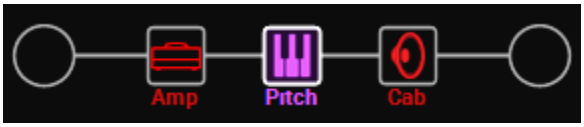
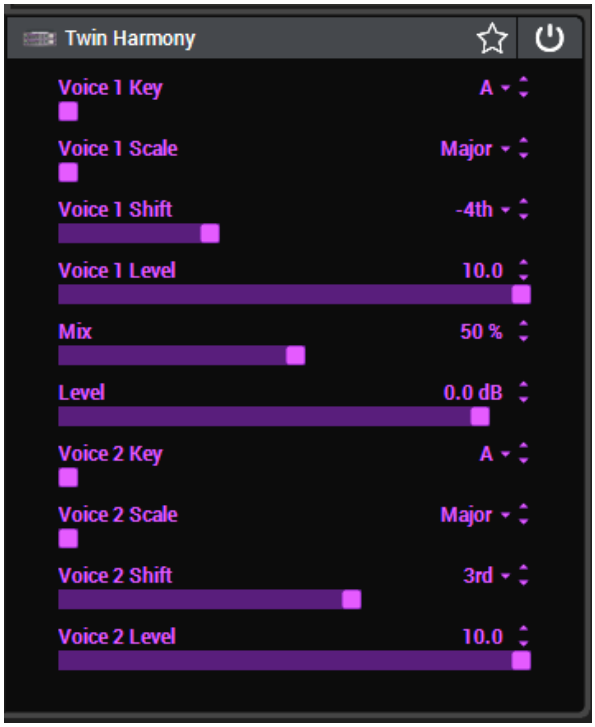


Figure 2.88 Setup for low tuning down 2 semitones.

Twin Harmony

This block creates one or two harmony lines from a single-note lead. With distorted sounds, insert this after distortion (e.g., amp or distortion effect). Inserting between an amp and cab often works best. Play cleanly for the best results.



Enter a Key and Scale, so that the Twin Harmony can generate harmonies that follow the scale. For example, to create a third in the key of E, Twin Harmony will generate a G# if the Scale parameter is Major, or a G if the Scale is Minor. The parameters are the same for the two harmonies:

- **Voice 1 (or 2) Key.** Specifies the song's key.
- **Voice 1 (or 2) Scale** selects among Major, Minor, Major Pentatonic, Minor Pentatonic, Harmonic Minor, Melodic Minor, Whole Tone, or Whole Diminished (see below).
- **Voice 1 (or 2) Shift** chooses the harmony's interval. 0 doubles the incoming note.
- **Voice 1 (or 2) Level** set the harmony line's level.

Tech Talk: Scales Used in the Twin Harmony

The following refers the scale notes to E. The intervals between notes are in parentheses.

Major: E • F# • G# • A • B • C# • D# • E (2-2-1-2-2-2-1)
Minor: E • F# • G • A • B • C • D • E (2-1-2-2-1-2-2)
Major Pentatonic: E • F# • G# • B • C# • E (2-2-3-2-3)
Minor Pentatonic: E • G • A • B • D • E (3-2-2-3-2)
Harmonic Minor: E • F# • G • A • B • C • D# • E (2-1-2-2-1-3-1)
Melodic Minor: E • F# • G • A • B • C# • D# • E (2-1-2-2-2-2-1)
Whole Tone: E • F# • G# • A# • C • D • E (2-2-2-2-2-2)
Whole Diminished: E • F# • G • A • A# • C • C# • D# • E (2-1-2-1-2-1-2-1)

Simple and Dual Pitch

These blocks generate parallel harmonies. The Dual Pitch has two sets of the same parameters as the Simple Pitch (fig. 2.89).

Figure 2.89 The Dual Pitch settings above create a major chord (the +4 interval is a major third, the +7 interval is a fifth). The parameters that control the harmony line levels have different names for the two blocks: Shift Level for the Simple Pitch, and Level 1 and Level 2 for the Dual Pitch's two harmony lines.

To refer to the Dual Pitch parameters, simply add 1 or 2 to the following Single Pitch parameters (except for Shift Level):

- **Interval** sets the harmony interval in semitones, from -24 semitones to +24 semitones.
- **Cents** fine-tunes the harmony interval in cents, from -50 to +50 cents.
- **Delay** sets the time between the incoming note and when the harmony plays, from 0.0 to 100 ms.
- **Shift Level** (called **V1 Level** or **V2 Level** in the Dual Pitch) sets the harmony line level.
- **Mix** chooses the percentage of dry sound and the harmony line(s).
- **Level** sets the block's overall level.

Applications

In addition to generating harmony lines, try the following:

- As with other pitch-shifting effects, this would usually go after distortion. However, you can create an octave-higher fuzz by setting Interval to +12, Cents to 0.00, Mix to 50%, and Delay and Level as desired. Follow this with distortion, like the Minotaur Distortion. The sound seems better-behaved, and sweeter, than with the Tycoctavia Fuzz. Also try Interval at -12.
- Although these effects are optimized for single-note lines, you can obtain chorusing effects with chords. Set Interval to 0, and then edit Cents for the desired chorusing effect. The Delay parameter can give this chorusing effect a different character.

Boctaver

This basic octave divider emulates a popular stompbox. Its clean sound complements the Legacy Sub Oct Fuzz, which is more about octave-based distortion.

The control set is simple:

- -1 Oct mixes in a sound one octave below the input signal.
- -2 Oct mixes in a sound two octaves below the input signal.
- Dry Level mixes in the original input signal.

When using Boctaver for single-note solos higher on the neck, the octave sound is essentially a sine wave. Lower notes have a bit more harmonic content, but the sound is still clean (fig. 2.90).

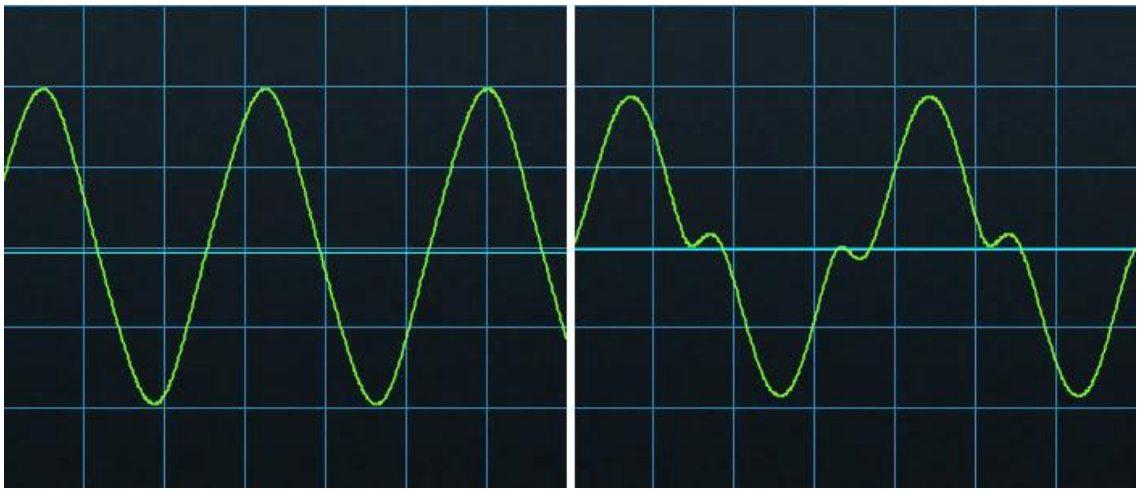


Figure 2.90 (Left) Dividing a 400 Hz sine wave by one octave retains the sine wave's shape. (Right) Changing the input frequency to 100 Hz still delivers a clean bass sound.

Here are tips on getting the most from Boctaver:

- Tracking is very good on fretted notes. Open strings tend to jump octaves as the note decays.
- The output level starts decreasing with frequencies approximately one octave above middle C.
- Playing two notes at the same time is kryptonite for this processor. Play cleanly, and if possible, mute any strings that aren't being played.
- As with most processors that track pitch, using the bass pickup and turning down its tone control improves tracking.

3 OSC Synth (Stereo Only)

The Synth tracks your guitar playing and generates synthesizer sounds. The tracking isn't great, but there are ways to improve it. The three oscillators have identical parameters, so to save space, fig. 2.91 shows only one set of Osc parameters. In the full interface, each Osc parameter is followed by 1, 2, or 3 for the three oscillators. Learn one, and you've learned them all.

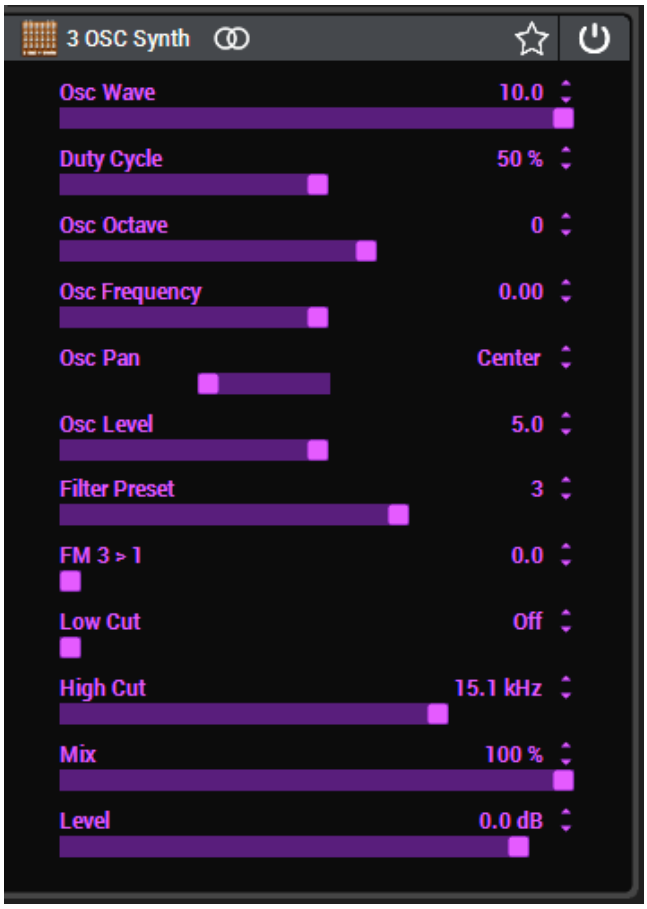


Figure 2.91 The Synth parameters are less complex than they might seem.

- **Osc Wave** morphs among three different waveforms. Fig. 2.92 shows what the waveform looks like at 0.0 (sine), 5.0 (triangle), and 10.0 (square, with ringing). Higher values generate more harmonic content.

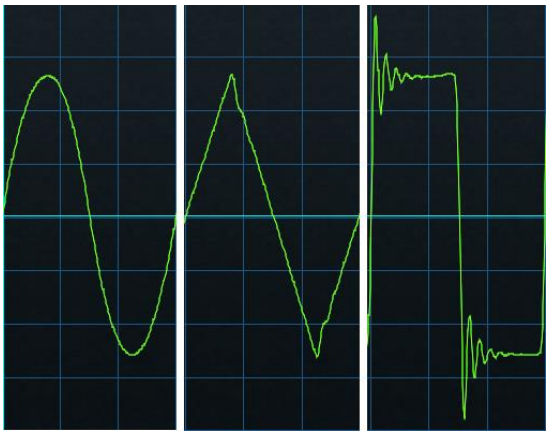


Figure 2.92 Left to right: Oscillator waveform at 0.0, 5.0, and 10.0. The Duty Cycle parameter is 5.0.

As the waveform transitions between 0.0 and 5.0, the sine wave looks more like a triangle wave. When the waveform transitions between 5.0 and 10.0, the triangle wave looks more like a square wave. Note that Filter Preset 3 is selected, which doesn't change the "pure" waveform as much as other Filter Presets (these add various resonances to the waveforms that alter the shape). Regardless of any differences, fig. 2.85 shows the baseline waveforms that the Filter Presets affect.

- **Duty Cycle.** We've met the concept before with LFOs (see Modulation Blocks), but audio waveforms can have different duty cycles too (fig. 2.93). The middle setting (5.0) has a 50% duty cycle. Changing the parameter more toward 0.0, or more toward 10.0, increases the harmonic content and the phase.

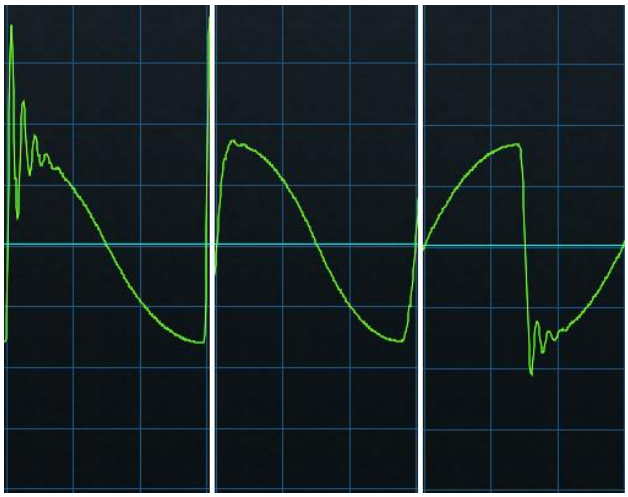


Figure 2.93 Sine wave changing duty cycle from 0.0%, to 17%, to 93%.

Tip: So far, this info is mostly for the people who've been wondering what the heck the Wave and Duty Cycle parameters do. From a musical perspective, all you really need to know is that changing Wave from 0.0 to 10.0 creates a "buzzier" sound with more harmonics, and moving Duty Cycle off center toward either extreme also increases the harmonic content.

- **Osc Octave** sets the oscillator pitch, from -3 octaves to +2 octaves.
- **Osc Frequency** fine-tunes the oscillator pitch in 100 cent increments, from -12.0 to +12.0 semitones.
- **Osc Pan** sets the oscillator sound's stereo position.
- **Osc Level** controls an individual oscillator's level in the mix of the three oscillators.
- **Filter Preset** has four options. 1, 2, and 4 apply a resonant filter with a decay to the note. Option 3 (my favorite setting) isn't about filtering but adds portamento (glide) between notes.
- **FM 3 > 1** sets the amount of Osc 3's output that's injected into Osc 1 to modulate Osc 1's frequency. This is the basis of *FM synthesis*, a type of synthesis that the Yamaha DX7 keyboard introduced to the mass market. The traditional way to use this parameter with a synthesizer would be to set Osc 3's Level to 0, turn up FM 3 > 1, and vary Osc 3's frequency with an Expression Pedal. But this isn't a synthesizer, so…anything goes.

We've already seen what **Low Cut**, **High Cut**, **Mix**, and **Level** do with other effects. These parameters work the same way in this effect.

So How Do I Get the Synth to Track My Playing?

Short answer: Manage your expectations. Long answer: It's possible to get decent tracking if you're willing to experiment a bit. The following applies to Filter Presets 1, 2, and 4 (Filter Preset 3 appears to follow its own tracking rules):

1. Insert the Horizon Gate before the Synth block.
2. Choose the Horizon Gate's **Extended** Range.
3. When starting out, play mostly in the 7th to 12th fret range. Use your neck pickup.
4. This is where it gets tricky. Reduce the Horizon Gate **Sensitivity** to where it feels like you're getting better tracking. 4.0 or 5.0 works for me *if* the input signal uses up the full Helix headroom. Lower Sensitivity settings can be better, but then you won't be able to trigger the Synth with high-pitched notes.
5. Here comes the plot twist: turn down your neck pickup's volume control. I don't know why this is important, it seems like editing the Sensitivity parameter should be all that's needed. But you'll (hopefully) find a magic setting that reduces the artifacts that normally happen at the end of synth notes. You may get better results by turning down the tone control instead of the volume control. Or, try a combination of the two.

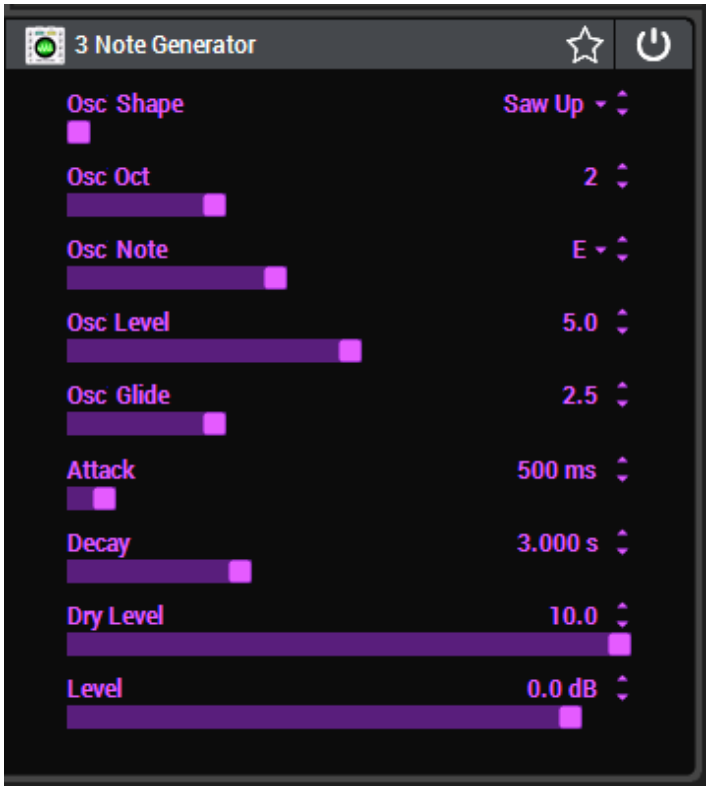
Because of the interaction between the Sensitivity parameter and the guitar's volume control, it's worth experimenting with both their settings to obtain the best tracking.

For synth bass parts, keeping my hand on the bridge and muting the strings *heavily* improved the tracking. But after bypassing the synth block, I was surprised to notice that the guitar notes were so muted they were just short blobs of sound—not sustaining notes. But hey, it works.

Granted, 3 OSC Synth won't make you sound like a 70s keyboard hero flailing away on a synthesizer—but you might sound like you found an old Korg X-911 guitar synth in someone's attic. For adding that special synthesizer line in one song, or for doubling bass on synth bass, this can work. The result depends mostly on how much you're willing to tweak the signal going into the Helix, and any processing that precedes the 3 OSC Synth, because that's what can coerce this quirky block into behaving.

3 Note Generator

This block plays three steady notes. It's for anyone who ever wanted to stomp on a footswitch and have a floor-rattling pedal-point bass, a background drone while you're playing, or an ambient background. With Snapshots, you can even call up different chords. The 3 Note Generator makes a sound only when enabled.



As with the 3 Osc Synth, to save space the Osc parameters in the above screenshot don't have numbers (1, 2, 3), because all three Oscillators share the same parameters:

- **Osc Shape** (fig. 2.94) chooses the note's waveform. A sine wave has no harmonics. Sawtooth waves have the most harmonics. A triangle wave is a sine wave with a little buzz, and a square wave has a "hollow" sound.

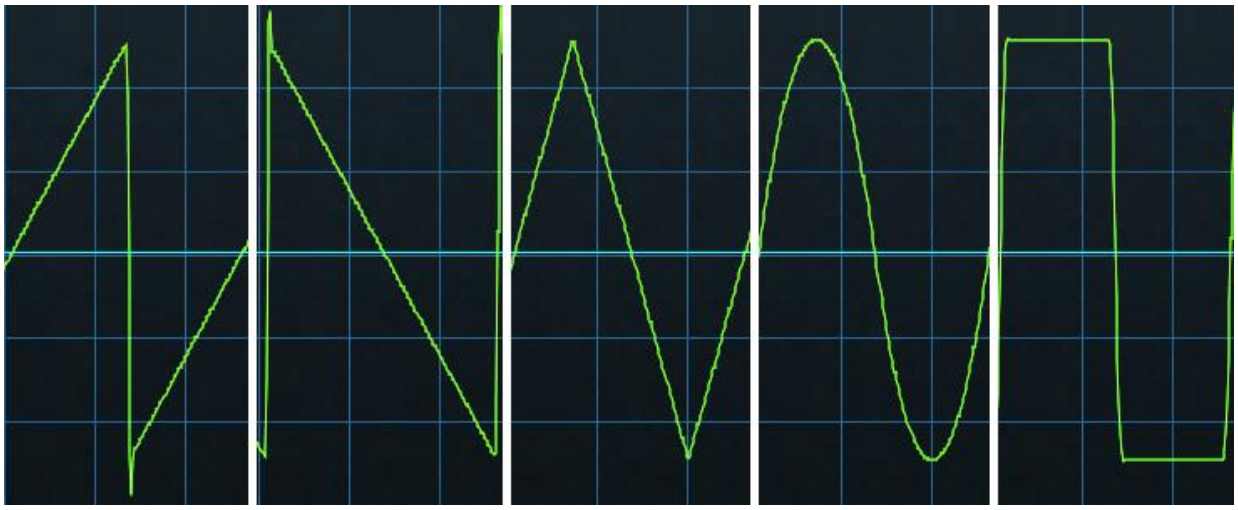


Figure 2.94 3 Note Generator waveforms. Left to right: Saw Up, Saw Down, Triangle, Sine, Square

- **Osc Oct** covers a 9-octave range (0 through 8)
- **Osc Note** chooses a note from C to B within the selected octave.
- **Osc Level** sets the contribution an oscillator makes to the final mixed output. Combining multiple waveforms can give complex, evolving waveforms (fig. 2.95).

Figure 2.95 Mixing multiple waveforms together produces a complex composite waveform.

- When changing note pitch, like when triggering Snapshots to choose different chords, **Glide** sets the time for a note to slide smoothly from its previous pitch to the new pitch.
- When you enable the 3 Note Generator block, **Attack** sets a time for the chord to fade in, from 10 ms to 10.000 seconds.
- When you bypass the 3 Note Generator block, **Decay** sets a time for the chord to fade out, from 10 ms to 10.000 seconds.

Tip: The fade-in time completes its entire time, even if you bypass the 3 Note Generator during the attack. If you bypass the 3 Note Generator while the decay is still happening and then re-enable the block, the attack will pick up from the current level set by the decay, and fade in over the amount of time that would normally be needed to go from the current level to the full level (fig. 2.96).

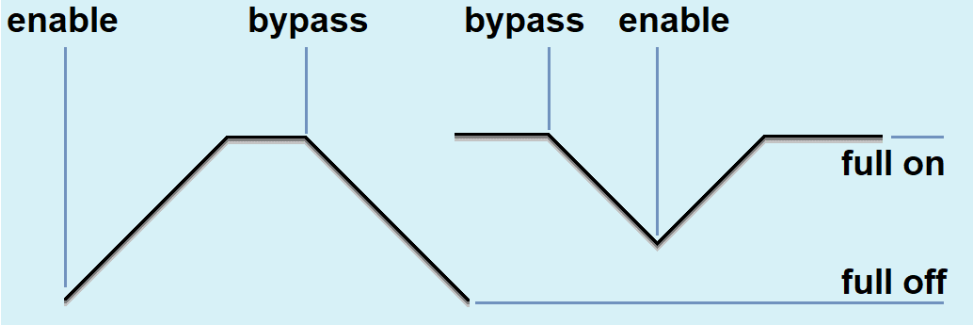


Figure 2.96 The Attack and Decay times are identical for the two examples. Left: Normal enable and bypass. Right: If you enable the block before a decay is complete, the attack picks up from the decay level, but takes less time because it doesn't start from full off.

- **Dry Level** allows mixing in your guitar with the 3 Note Generator sound.
- **Level** sets the block's output level.

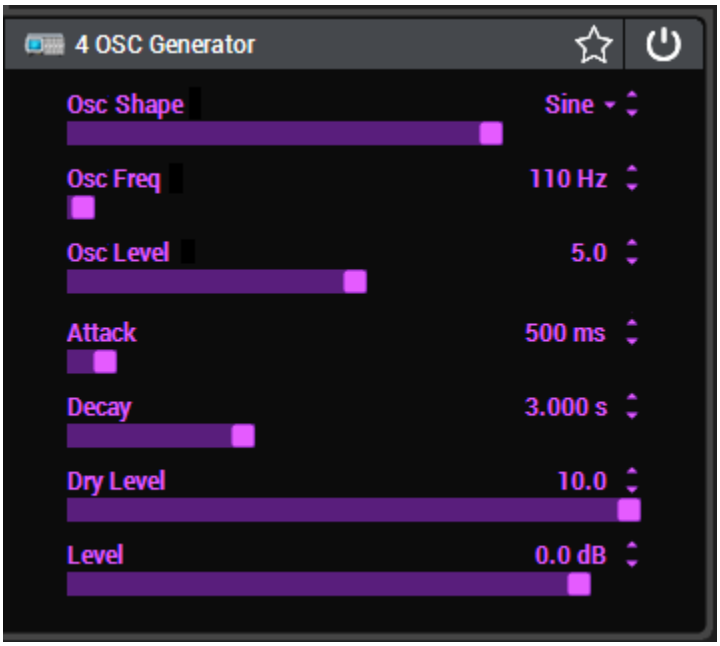
Tip: Insert the 3 Note Generator in a parallel path, and then the other path can process the guitar. The E Drone Raga and Hindustani presets use this approach.

4 OSC Generator

This is conceptually like the 3 Note Generator. You can bring in a background sound by enabling the block, and have the sound decay over time after bypassing it. However, there are three significant differences:

- There are four oscillators instead of three.
- Pitches are not quantized into octaves and semitones. Instead, they're variable from 20 Hz to 10 kHz.
- There's no Glide parameter.

As with the 3 Note Generator, to save space the Osc parameters in the screenshot below don't have numbers (1, 2, 3, 4), because all four Oscillators share the same parameters. Aside from the continuously variable **Osc Freq**, the parameter functionality is identical to the 3 Note Generator.



Tech Talk: Helix's Test Equipment

Aside from musical uses, the 4 OSC Generator is a useful piece of test equipment because it can generate various waveforms for testing. Many DAWs have diagnostic metering (like Studio One's Spectrum Analyzer and Oscilloscope plug-ins, which provided many of the images in this book). Having a predictable source of waveforms with varying harmonic content can analyze frequency response, filter resonances, and more.

Filter Blocks

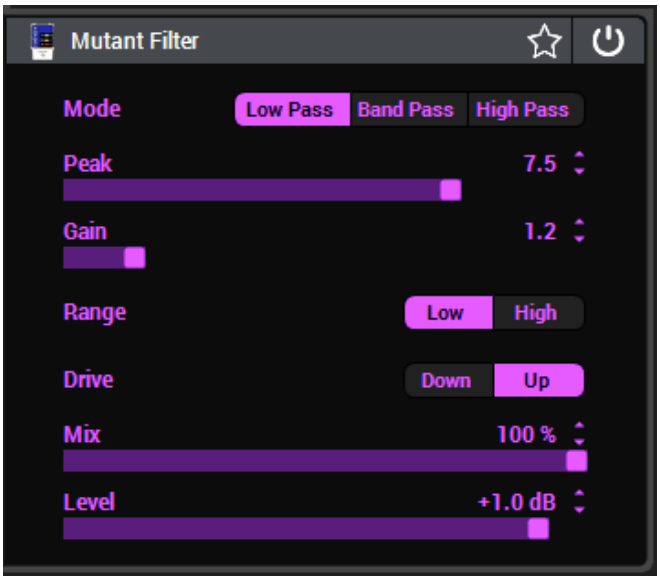
The EQ blocks change tone statically. In other words, they're "set-and-forget," unless you want to vary one of the parameters with a pedal. The Filter block effects change tone dynamically. Three of them respond to the dynamics of your playing.

Controlling filters with your dynamics can be tricky, because your playing needs to cover a consistent dynamic range to obtain consistent results. Restricting the dynamic range of the signal feeding the filter can make the tracking more compatible with your playing technique. The simplest solution is inserting a compressor before the filter. The Deluxe Compressor is a good option, as is the LA Studio Comp in Limit mode.

The filters in these effects can produce resonant peaks. If they overload subsequent stages, inserting a dynamics processor immediately after the filter can reduce the peaks' levels. The LA Studio Comp in Limit mode is a good choice, but other compressors will also work.

Mutant Filter

The Mutant Filter is like a wah that tracks the dynamics of your playing. The parameters in the screenshot are set for a typical autowah effect.



Playing harder can push the filter frequency higher or lower, depending on the parameter settings:

- **Mode.** Options are Low Pass, Band Pass, and High Pass (fig. 2.97). All three responses have a prominent peak. The Lowpass includes frequencies below the peak, Bandpass suppresses frequencies on either side of the peak, and Highpass includes frequencies above the peak. The peak's frequency moves higher or lower depending on the incoming level, but the filter response's shape doesn't change much at different frequencies.

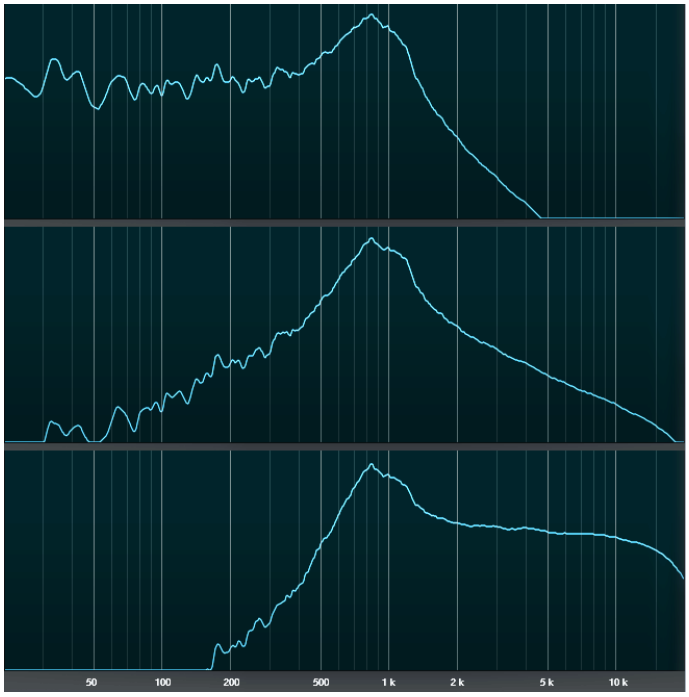


Figure 2.97 Top to bottom: Low Pass, Band Pass, and High Pass filter responses, with Peak close to maximum.

- **Peak** is the filter's resonance control. Higher values give a sharper sound.
- **Gain** sets the filter tracking's sensitivity to your playing. With high values, even extremely light picking will move the peak over a wide frequency range.
- **Range** causes the frequency to change over a lower or higher range of frequencies. Low's range is about 200 Hz to 2 kHz. High's range is roughly 350 Hz to 5 kHz.
- With **Drive** set to Up, playing harder pushes the filter frequency higher. With the Down setting, playing harder pushes the filter frequency lower.
- **Mix** sets the blend of dry and filtered sound.
- **Level** determines the output level.

Adjusting the Sensitivity

Whether the Mutant Filter responds satisfactorily to the dynamics of your playing depends on several factors. The first, and most important, is a consistent playing style with respect to dynamics. That said, I find that even with Gain at 0.0, the Mutant Filter is too sensitive—in Up mode, playing with only medium force kicks the filter to its highest frequency. However, in Down mode the sensitivity is less crucial. If you encounter issues, here are some tips on how to tailor the response to your playing:

- Turn down your guitar's volume control, if the Mutant Filter is the first effect in a chain.
- If the Mutant Filter is further down a chain, add a Volume (Gain) block before the filter, and reduce the level going into it. Depending on how much you need to reduce the incoming level, you'll likely need to increase the Mutant's Level parameter. If that isn't enough to compensate, you may also need to boost the level after the filter, with another effect's Level parameter, or using a Vol Gain block.
- As mentioned previously, insert a compressor before the Mutant Filter, like the LA Studio Comp or Deluxe Compressor, and reduce the dynamic range to which the filter responds.
- Use the Autofilter block instead of the Mutant Filter. This Line 6 original is less fussy and has a more comprehensive set of editable parameters.

Mystery Filter

This filter tracks your playing, but it's different from other envelope followers. It dips down from a high to low cutoff frequency, then rises back up again.

The screenshot shows settings that sound good with my playing, but the mystery is whether they'll work well for you. The Sensitivity parameter is super-touchy, and the Attack and Release parameters interact with it. Here's what the parameters do:

- **Sensitivity.** Lower values require hitting the strings harder to push the filter frequency down. Higher values are more sensitive to your dynamics.
- **Frequency** set the starting filter frequency from which the filter sweeps down. Higher settings start from a higher frequency.
- **Resonance** determines the filter's sharpness.

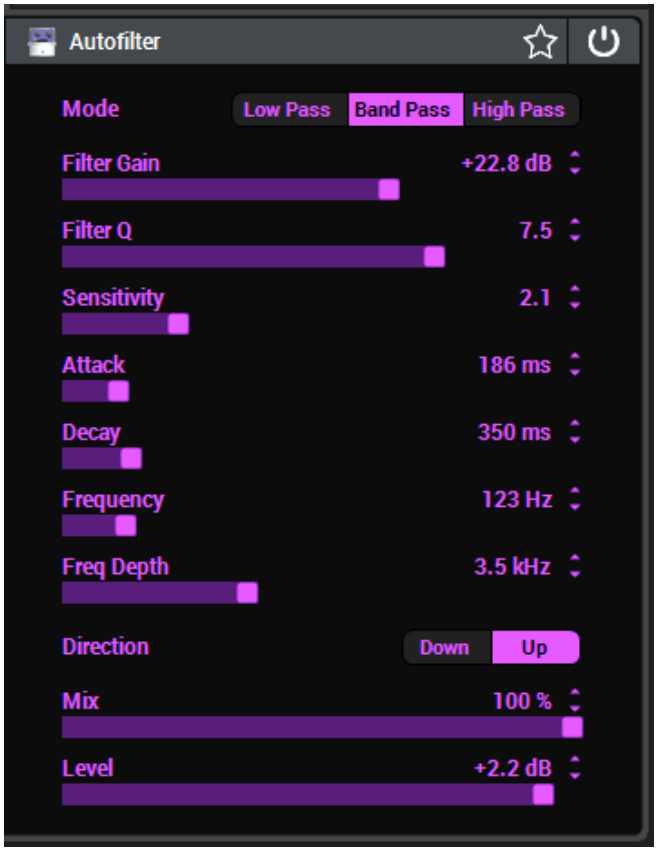
Tip: Set Resonance to maximum when you're first trying out the Mystery Filter effect. This will make changes in tracking more obvious as you adjust the various parameters.

- **Attack.** Lower values cause a slower downward sweep when first triggered. Attack interacts with Sensitivity—when playing softly, the attack time is longer, even with higher Attack values.
- **Release.** Lower values smooth the transition from the lowest filter frequency back up to the original starting frequency.

Tip: The parameter settings are quite fussy. If you find settings that work well with your guitar and playing style, save the block as a Favorite.

Autofilter

This is like the Mutant Filter, but it adapts more easily to your dynamics and has more editable parameters.



- **Mode.** Options are Low Pass, Band Pass, and High Pass. The response curves are the same as in fig. 2.97 for the Mutant Filter.
- **Filter Gain** adjusts the level through the filter. Higher values give a more intense filter effect.
- **Filter Q** controls the filter resonance. Higher values give a sharper sound.
- **Sensitivity** determines how the filter frequency tracks your dynamics. Higher values cover the widest frequency range and respond to less forceful picking. Lower values make the tracking less sensitive to your dynamics, i.e., you have to play harder for the filter to reach higher frequencies.
- After you play a note or chord, **Attack** sets the time for the filter to go from the initial frequency to the maximum frequency (as determined by the Sensitivity parameter and your playing's dynamics). Note that low settings can cause a "warble" as the filter tries to follow rapid level variations. Increase the Attack time to eliminate the warble.
- As the input level goes lower, **Decay** sets the time for the filter to fall back to a lower frequency.
- **Frequency** sets the filter's lowest frequency, i.e., the frequency when there's no input signal.
- **Freq Depth** sets the maximum filter excursion.
- With an Up **Direction**, playing harder pushes the filter frequency higher. With the Down setting, playing harder pushes the filter frequency lower.
- **Mix** sets the blend of dry and filtered sound.
- **Level** determines the output level.

Asheville Pattrn

This effect is useful for bass as well as guitar—see Rhythmic Patterns for Bass in the chapter on Helix Bass Processing. Either an LFO or sequencer steps through 8 filters (fig. 2.98), each with an adjustable level. Level 1 sets the level of a lowpass filter. Levels 2-8 are bandpass filters, each tuned to a successively higher frequency. When learning this effect, turn up the Drive parameter to accentuate the highs, which emphasizes the effect of the filters. Rather than simply describe the parameters, we'll cover how to learn what the effect does.

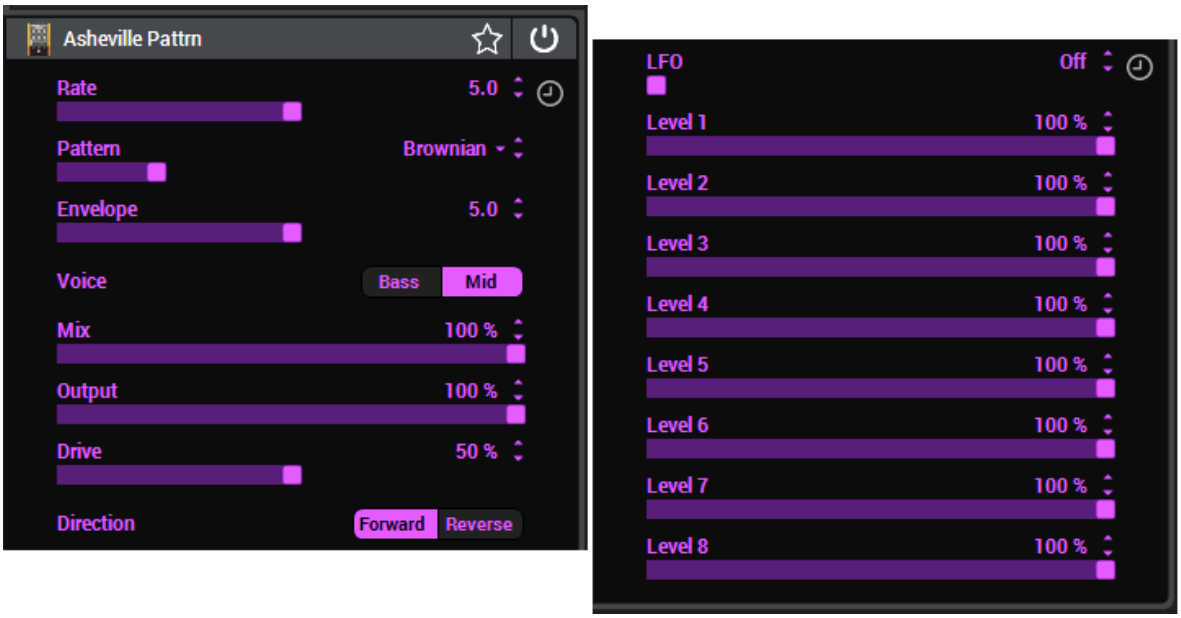


Figure 2.98 The Pattern generation section is on the left. The LFO and filter levels are on the right.

The LFO Section

Set the **Pattern** parameter Off to use the LFO. The **Rate** (or Note Sync, if enabled), **Envelope**, and **Direction** parameters have no effect. The **LFO** parameter sets the LFO rate. The rate can also sync to tempo.

Set the **LFO** parameter to around 50%, so its effect is obvious. Start with **Level 1** at 100%, and **Levels 2-8** at 0%. Play a sustained power chord. Bring various filters up to 100%, and then back down to 0%. You'll hear that each filter has its own effect on the sound. To achieve mild phase shifter effects, bring up some filters, but not others—for example, bring up Levels 3, 5, and 7 to 100%, while leaving 2, 4, and 6 at zero.

Using Patterns

Set the **LFO** parameter to Off. Choose **Stair** for the **Pattern**, **Rate** around 5.0 to 6.0 (with Note Sync, try 1/8th notes), and **Envelope** around 2.5. Turn up all the filter level controls. You'll hear a higher filter frequency on each successive Pattern step, until the Pattern jumps back to the beginning. Now, vary the **Envelope** parameter. Lower values give a more percussive envelope decay, while higher values add more of an attack. This is a fun parameter to assign to an Expression Pedal.

Next, audition the 20 Patterns. The patterns step through the filters in different orders. The **Direction** parameter reverses the pattern direction—for example, the Stair pattern goes from the high filter to the low filter.

Note that when synched to tempo, Asheville Pattern doesn't always start on the beat. If necessary, hit the tempo footswitch briefly (or send Helix a MIDI controller 64 message) on the beat to reset the pattern.

Wah Blocks

Most wahs use a *bandpass* filter, which creates a peak at the filter's *resonant frequency*. The filter frequency is variable between a low and high frequency. Sweeping between these with the Expression Pedal creates the wah effect. Wah blocks default to Expression Pedal 1.

There are two main Helix Wah designs (fig. 2.99). The UK Wah 846, Teardrop 310, and Teardrop Bass Q have fixed low and high frequencies (the Teardrop Bass Q also has a Q Time control for shaping the resonance). However, when controlling these wahs from a MIDI controller, you can set Min and Max positions in the Controller Assign section. The remaining eight wahs have adjustable low and high frequencies. The low frequency can range from 20 Hz to 500 Hz, and the high frequency, from 500 Hz to 5 kHz.

Figure 2.99 Left: Wah with fixed low and high filter frequencies. Right: Wah with editable low (FcLow) and high (FcHigh) filter frequencies.

The UK Wah 846's lowest frequency is about 700 Hz. Its highest frequency is around 1.65 kHz, so the filter's range is a little over an octave (fig. 2.100). The Teardrop 310 covers from about 500 Hz to 1.65 kHz, so its range spreads a little over an octave-and-a-half.

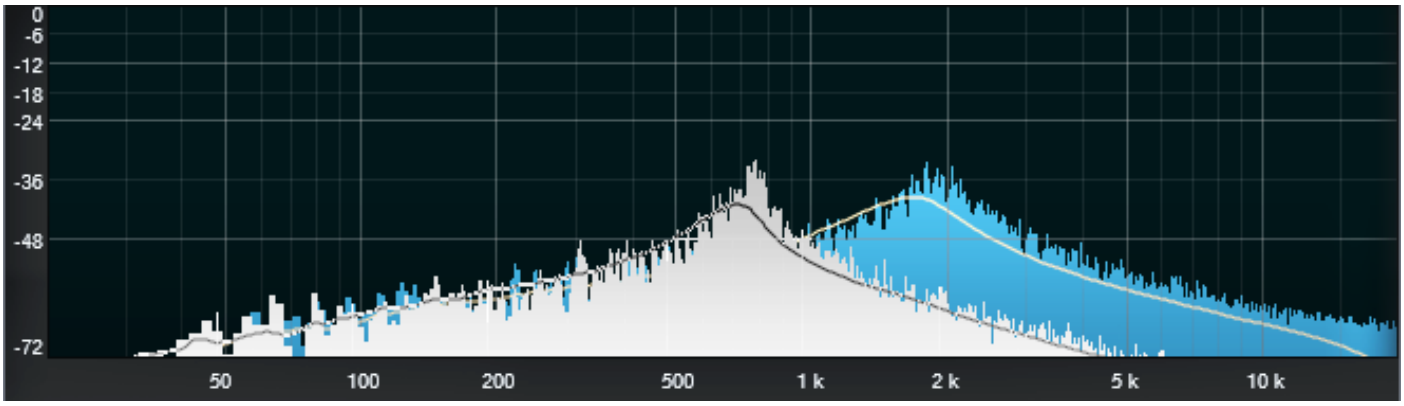


Figure 2.100 The gray spectrum shows the UK Wah 846's response at the heel (low frequency) position. The blue spectrum shows the response at the toe (high frequency) position.

Setting the same low and high frequencies with different Wahs won't necessarily sound the same, because their bandpass filters have different characteristics. For example, the filter peak may be more or less resonant, or the *bandwidth* might be narrower or broader (fig. 2.101). Wahs can also have different peaks and bandwidths at

their low and high frequencies. The Chrome Custom wah has a wider bandwidth at its highest frequency than at its lowest frequency.

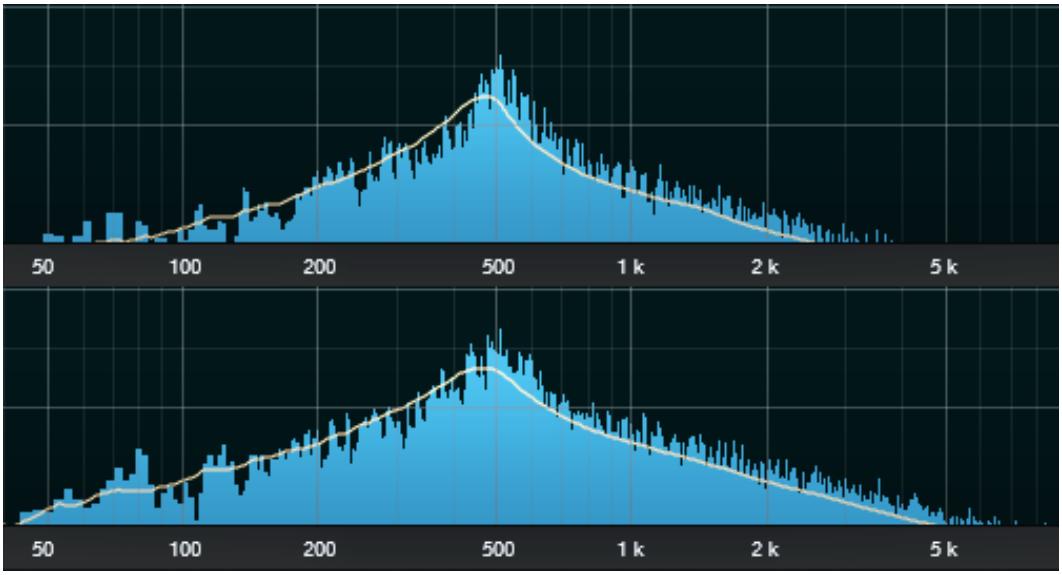


Figure 2.101 Top: Chrome Custom wah at 500 Hz. Bottom: Vetta Wah at 500 Hz. The Chrome Custom has a more resonant peak, and a narrower bandwidth.

Teardrop Bass Q (1.5)

The Teardrop Bass Q frequency response differs from other wahs. Fig. 2.102 shows the low and high peaks, with maximum Q. Note that it doesn't attenuate the bass frequencies much. The lowest wah frequency boosts the response around 200 Hz, creates a deep midrange scoop, and retains some high frequencies. The highest wah frequency creates a peak in the upper midrange, retains the low frequencies, and rolls off the highs.

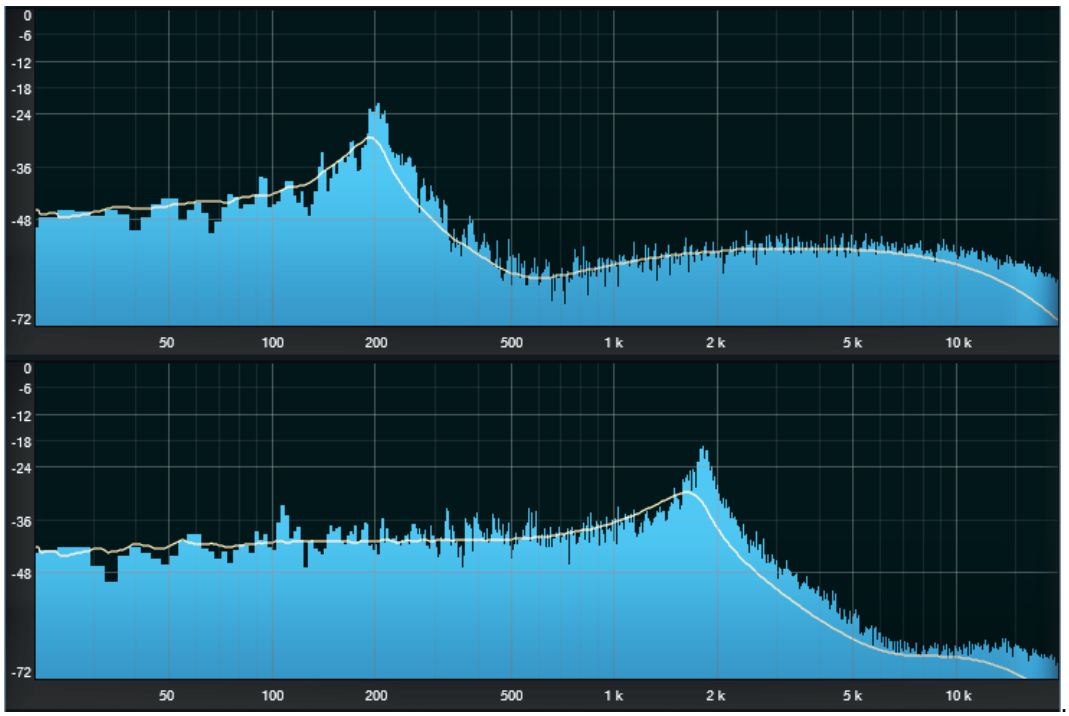


Figure 2.102 Response curves for the Teardrop Bass Q at the lowest and highest frequencies. Q Trim is at maximum.

Although considered a bass effect, the Teardrop Bass Q provides a different kind of wah character with guitar that can be more subtle than a conventional wah. For example, pull the Q Trim down, set Mix to around 50%, and the wah response feels more like an unusual tone control than a wah.

The Cocked Wah Sound

The traditional cocked wah sound occurs by inserting a wah before an amp, setting the pedal to a stationary position, and not moving the pedal. The term comes from drawing back a firearm's hammer so that it's set for firing, but leaving the hammer drawn back ("cocked and ready"). The wah provides a frequency boost that changes the amp's tone in one or more ways:

- Emphasizes the frequencies being boosted.
- Causes an amp on the verge of breakup to go into distortion sooner at the wah's resonant frequency.
- Offsets a scooped amp's frequency response somewhat by boosting the midrange.

The cocked wah effect may dominate the sound when inserted before a high-gain amp, because of excessive distortion. There are two main ways to lessen the effect's intensity:

- Reduce the amp's drive.
- Use a wah whose filter isn't as resonant. To create a low-resonance wah preset for this application, insert a Parametric EQ in parallel with a dry path, and invert the dry path's phase. See the section Customizable DIY Wah for more information about this effect.

Utility Blocks

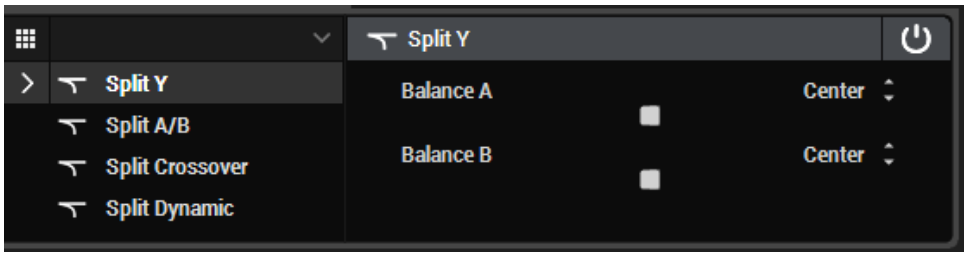
Line 6 doesn't use the term "utility blocks," but I created a category to cover several blocks that aren't particularly glamorous—yet are extremely useful.

Input Split Block

This appears only with a parallel path. It provides four ways to split the input signal to the parallel paths.

Tip: You can bypass the Split block. When bypassed, the left and right channels go to both paths, without modification.

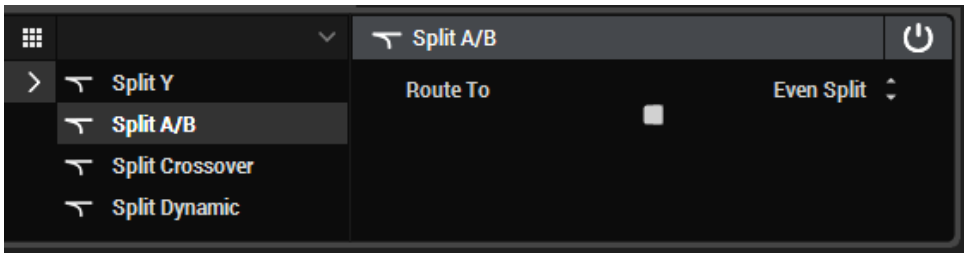
Split Y



Y sends the signal equally to two parallel paths. Balance A adjusts the stereo balance of the audio going to Path A, while Balance B does the same for Path B. Here are some examples on how to use two parallel paths:

- Feed two amps with complementary sounds. When combined back together in mono, the sound is bigger than either amp by itself.
- Create a big stereo image by having one amp in one path, and a different amp in the other path.
- Create EDM-friendly stereo imaging by inserting tempo-synched effects that run at different rates in the two paths.
- With bass, run the dry bass sound in one path, then insert any effects in parallel. Then no matter what effects you add, the bass will always retain its bottom end.

Split A/B



This lets you change the ratio of the signal going to each path. Typical applications:

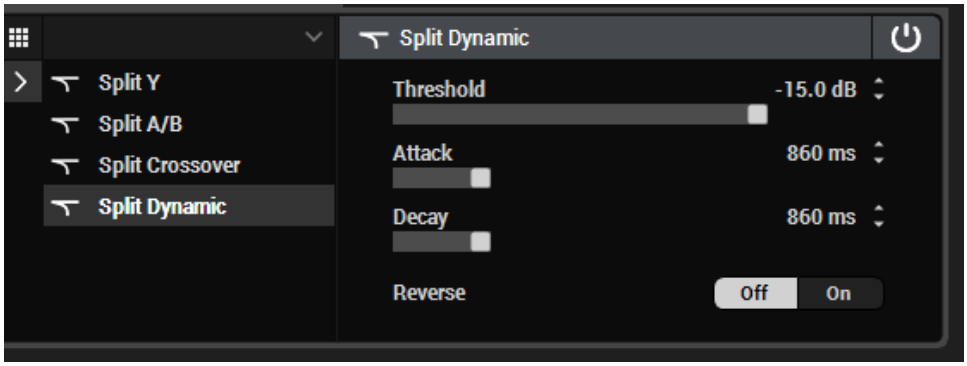
- Assign the A/B split to an Expression Pedal, thus sending the guitar to two different sounds you can blend however you want. With a mono output, this lets you "morph" between two different sounds.
- Set up a rhythm sound in one path and a lead sound in the other, then assign an Expression Pedal to the split so you can transition smoothly between the two sounds.

Tip: Because all of Helix Floor/LT/Native's four paths are stereo, A/B split sends the left and right signals to both stereo paths.

Crossover

The Crossover makes it easy to do bi-amping, where you optimize two amps for high and low frequencies. Chapter 5 is dedicated to bi-amping applications. This chapter includes more information about using the Input Split block for optimum bi-amping.

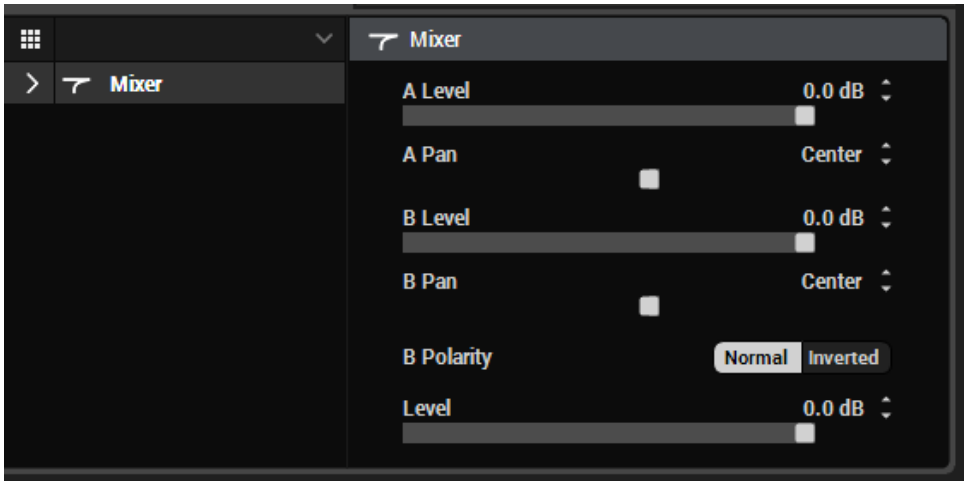
Dynamic



- **Threshold** determines how this split reacts to the signal level. Audio below the Threshold goes to Path A, audio above the Threshold goes to Path B.
- **Attack** sets the time for the signal to go to Path B once the level exceeds the Threshold.
- **Decay** sets the time for the signal to return to Path A after going below the threshold.
- **Reverse** changes how the split sends the signal. Audio above the Threshold goes to Path B, audio below the Threshold goes to Path A.

Mixer Block

This appears only with a parallel path. The mix provides different ways to combine the parallel paths back into a single output.



- **A Level** sets the A Path's level.
- **A Pan** places the A path in the stereo field. With mono audio, the signal goes anywhere between the left and right channels. With stereo audio, A Pan acts as a balance control. Moving the slider to the left increases the left channel's level while reducing the right channel's level. Moving the slider to the right increases the right channel's level while reducing the left channel's level.
- **B Level** sets the B Path's level.
- **B Pan** affects the B Path the same way A Pan affects the A Path.
- **B Polarity** inverts the B path's polarity (also called phase) compared to the A Path.
- **Level** is the Mixer's master output level control. It affects the output of both paths equally.

The Volume/Pan Block

It's not particularly exciting compared to some cool delay, but this is one of the most useful Helix blocks.

Mono and Stereo Operation

The Volume and Gain blocks (fig. 2.103) are available for either stereo or mono presets.

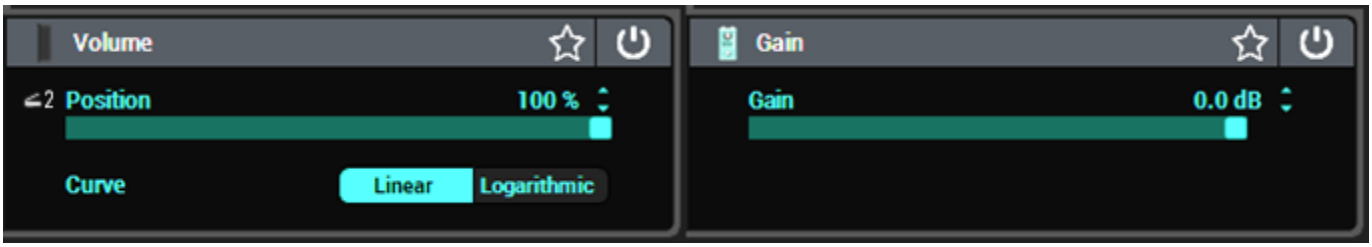


Figure 2.103 Left: Volume block. Right: Gain block.

The Volume block links to Expression Pedal 2 by default, and has two parameters:

- **Position** displays volume based on the pedal position, from 0 to 100%. However, you can change the default range with the Controller Assign function, by editing the Min Value and Max Value. For example, if you set the Max Value to 50%, then the pedal's full travel goes from 0% to 50%, not 0% to 100%. This allows different presets, or paths within a preset, to respond differently to pedal movement.
- **Curve** determines the pedal's "feel," based on how the human ear responds to level (fig. 2.104).

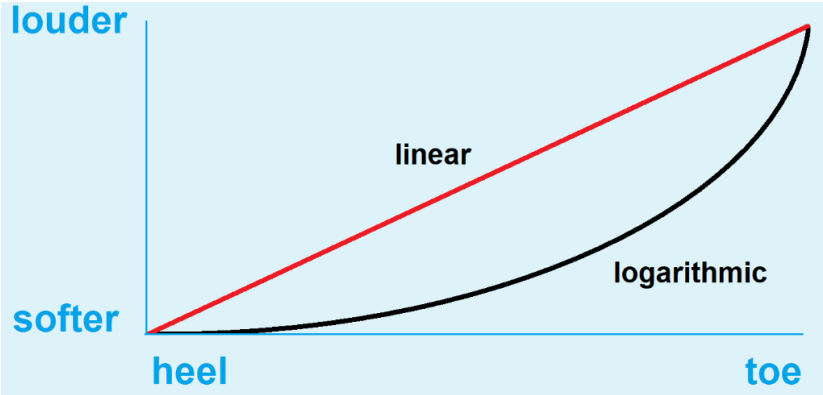


Figure 2.104 The Logarithmic response more closely tracks how the human ear responds to level changes.

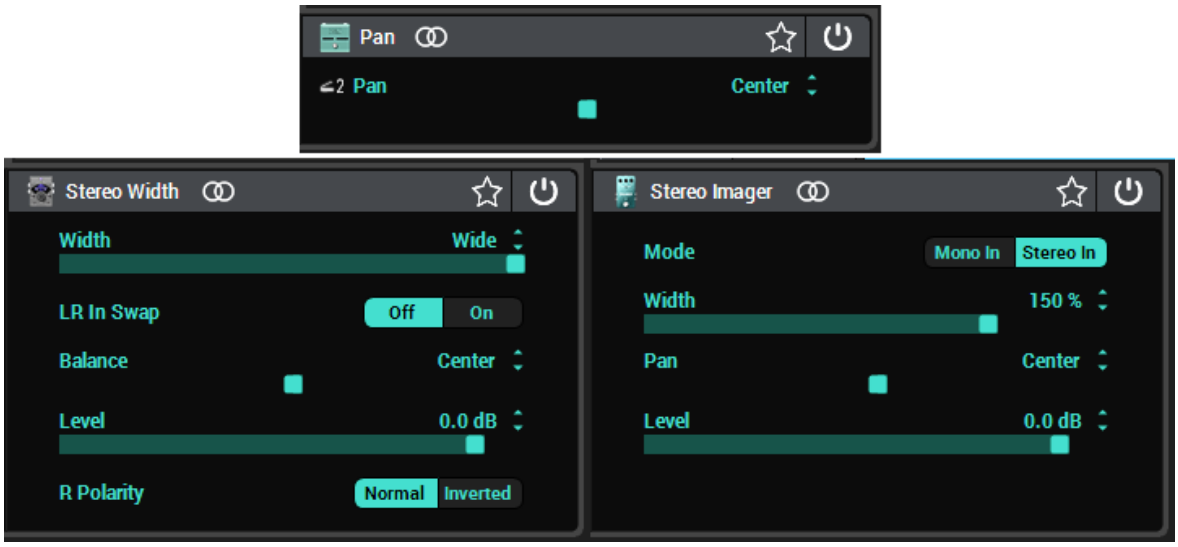
> For altering volume, a Linear curve causes the most perceived level change during the initial range of the pedal's travel. Exponential distributes the level change more evenly over the pedal travel. Normally, you'd select Exponential. However, when using an external foot pedal with a logarithmic taper to control volume, then the Curve should be linear, so you don't "stack" two logarithmic responses.

If all your presets have a Volume block, then you might want to link the Expression Pedal globally. In the Global settings, choose the Exp Pedals submenu, and turn the EXP 2 Position knob to "Global." This prevents a situation where upon loading a preset, the Expression Pedal's physical position differs from the programmed position.

The **Gain** block has a single parameter that varies gain between -120 dB and +12 dB.

Stereo-Only Blocks

Pan, Stereo Width, and Stereo Image blocks are available only with Volume/Pan in stereo mode. They control various aspects of the stereo image. The Pan position defaults to Expression Pedal 2.



Pan

- **Pan** places a *mono* signal anywhere in the stereo field, from left, through center, to right.
- With a *stereo* signal, Pan is a balance control. At center, both channel levels are equal. Moving Pan further to the left increases the left channel level, while reducing the right channel level. At full left, you hear only the left channel. Moving Pan further to the right increases the right channel level, while reducing the left channel level. At full right, you hear only the right channel.

Stereo Width

- The **Width** parameter collapses stereo audio to mono or spreads it to the full stereo width (fig. 2.104).

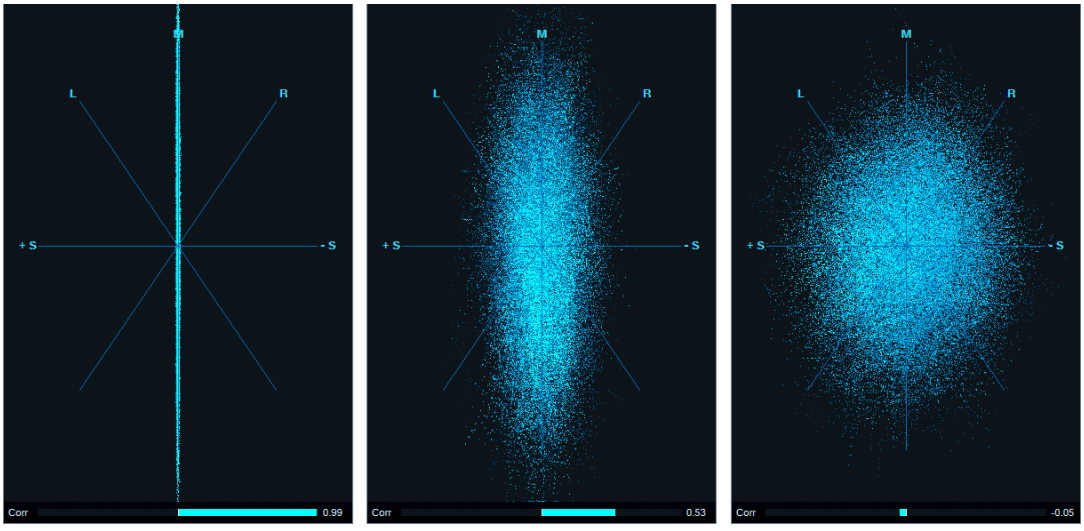


Figure 2.105 Width settings, left to right: 3%, 60%, 100%.

Fig. 2.105 shows level vertically, and stereo spread horizontally. The left-most image is essentially mono. The level builds up in the center compared to the next two images. This illustrates center-channel buildup, where collapsing a stereo signal to mono adds the right and left channels together, which increases the center's level. The middle image shows more of a spread to the left and right sides, which brings down the center level a bit. The right-most image shows that the audio's energy is now distributed equally to the left and right channels, so the audio no longer builds up in the center.

- **LR In Swap** switches the left and right channels. What was left becomes right, and vice versa.
- **Balance** works the same way as the Pan control in its stereo mode. At center, both channel levels are equal. Moving Balance further toward a channel increases that channel's level and decreases the other channel's level.
- **Level** sets the block's output level.
- In the Invert position, **R Polarity** flips the right channel's polarity (commonly called "phase").

Tip: To check whether a stereo preset will sound acceptable in mono, place this block at the end of an effects chain. Set **R Polarity** to Inverted, and listen for any problems, like a thinner sound.

Stereo Imager

- **Mode** chooses between a mono or stereo input.
- **Width.** With a mono input, the stereo imager uses a widening process that, if the signal collapses back to mono, creates comb filtering (fig. 2.106). To maintain a dramatic stereo separation, the signal must remain in stereo. For example, if you widen the signal and play it back through a mono radio, the sound will be thin. Some would also advise against this even for live use, although that's much less fraught.

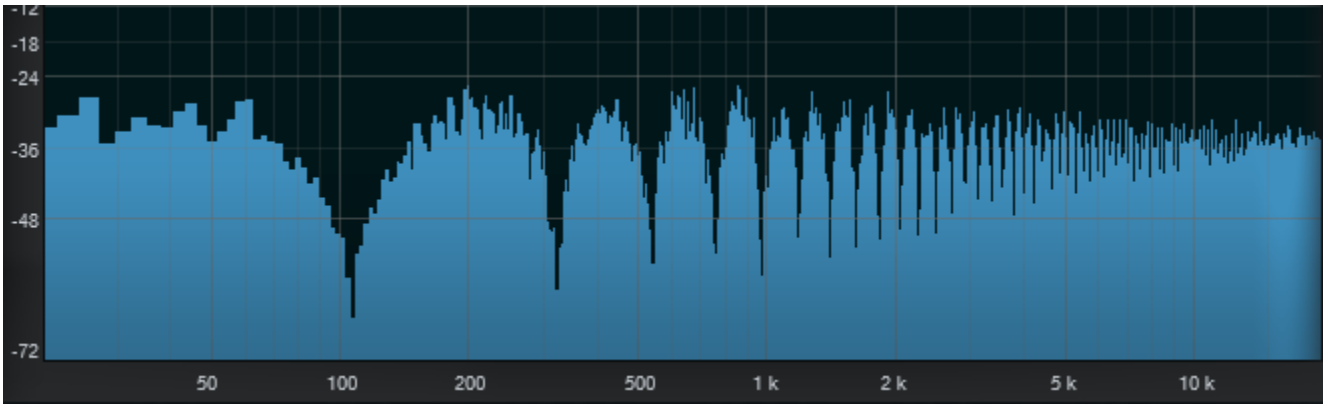


Figure 2.106 Combining the synthesized stereo back into mono creates comb filtering, which leads to a thinner sound.

With stereo, width at 100% plays back the stereo signal as if it wasn't processed. 0% collapses it to mono. 200% spreads the right and left channels out further at the expense of less center. Collapsing the widened signal back to mono reduces the level but doesn't alter the tone significantly.

- **Pan** acts like a Balance control, as described in the section on Pan.
- **Level** sets the block's output level.

Send/Return and FX Loop Blocks

The Send/Return and FX blocks interface Helix hardware's Send and Return/Aux In jacks with external devices—stomp boxes, amps, studio hardware, and so on. Helix connects to these kinds of hardware devices with patch cords. Different Helix models have different Send, Return, and Aux In configurations.

Model	Send Jack	Return Jack	Aux In Jack
Helix Floor	**4** (mono)	**4** (mono)	**1** (mono)
Helix LT	**2** (mono)	**2** (mono)	-
Helix Rack	**4** (mono)	**4** (mono)	**1** (mono)
HX Stomp	**1** (stereo, L and R)	**2** (mono)	-
HX Stomp XL	**1** (stereo, L and R)	**2** (mono)	-
HX Effects	**2** (mono)	**2** (mono)	-

For example, Helix Floor has 4 mono send jacks and 4 mono return jacks. When used as effects loops, these can connect to up to:

- Four mono-in/mono-out effects
- Two stereo-in/stereo-out effects
- One mono in/stereo out effect and two mono-in/mono-out effects

As another example, HX Stomp has a stereo Send jack, but plugging a mono guitar cord in all the way turns it into a mono Send. So:

- HX Stomp can natively do mono-in/stereo-out.
- Plugging a stereo-to-dual-mono adapter cable into the stereo send jack allows using two mono-in/mono-out effects.

Any Helix can do a 4-cable setup with amps. Some Helix hardware models can even use Send connections to create surround and multi-channel audio setups.

Note that different external devices expect different input/output levels. *Instrument levels* are the relatively low-level signals produced by guitars and basses with passive pickups. Active pickups usually have the same levels, for similar compatibility with guitar amps. Stompbox inputs and outputs are designed for instrument levels.

Line levels are higher-level signals. They're generated by electronic instruments, like keyboards, drums, and mixer outputs. Studio hardware rack-mount processors have inputs and outputs designed for line-level signals.

Fortunately, under *Global Settings* > *Ins/Outs*, you can specify whether each send/return pair is compatible with line or instrument levels. For more information on setting levels and inputs/outputs for your specific model, please refer to its manual.

I/O Selection

Different Helix hardware presents different send, return, and loop configurations for mono or stereo I/O. Fig. 2.107 shows the mono and stereo options for HX Stomp.

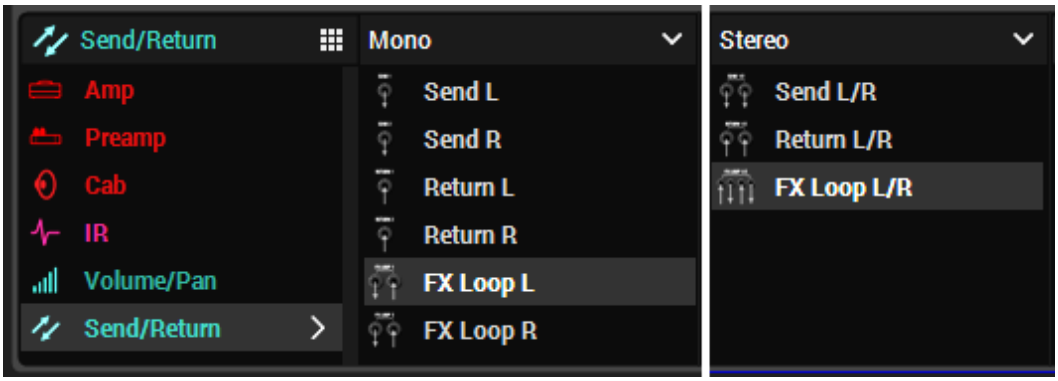


Figure 2.107 Mono and Stereo send/return configurations for HX Stomp

Helix Floor has many more options, due to the expanded I/O (fig. 2.108).

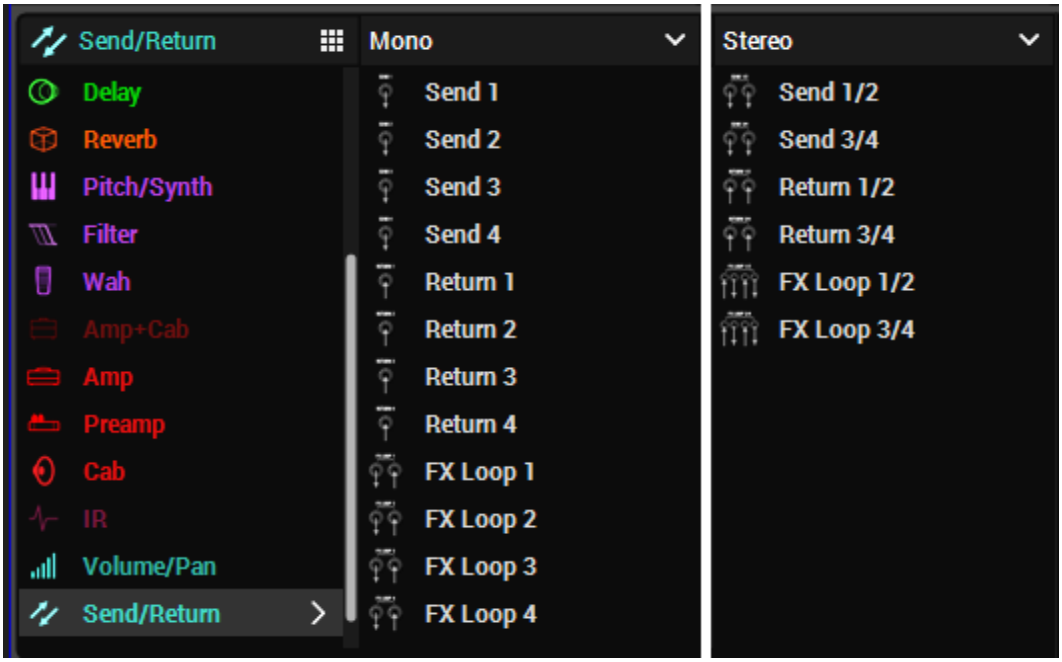


Figure 2.108 Choosing I/O with Helix Floor.

FX Loop Block

An *FX Loop* block, which can be mono or stereo, inserts within a Helix signal path like any other effect. Functionally, it accesses the effects loop's Send and Return connections from a single block. Fig. 2.109 shows how an FX Loop can add a series effect from external hardware into a Helix FX preset.

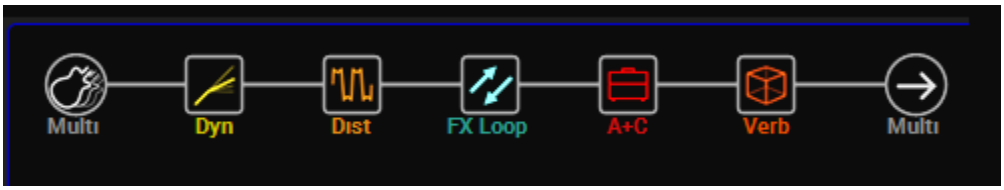


Figure 2.109 The FX Loop is inserting a hardware effect between the Helix's Distortion and Amp+Cab blocks.

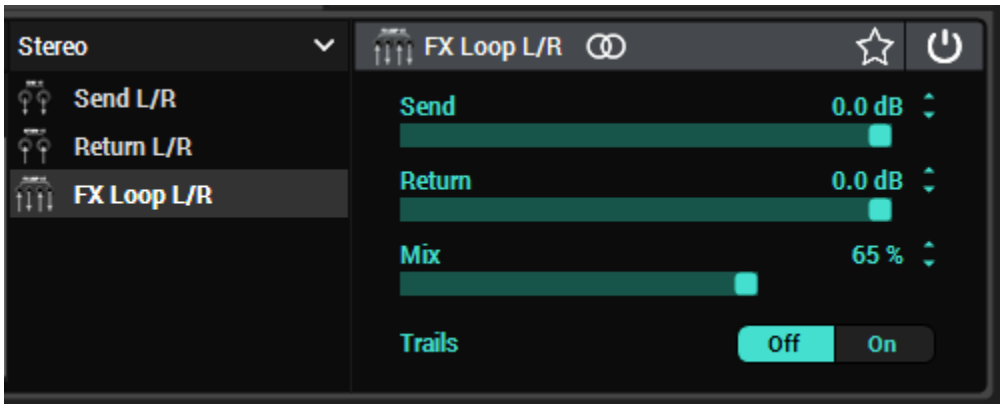
To insert a mono effect into a path, the FX Loop sends audio to a Send jack (e.g., Send 1). This jack patches to the external effect's input. The external effects' output patches to a Return jack (e.g., Return 1). This completes

the loop. Similarly, with Helix floor, if you used FX Loop 2, 3, or 4, it would send audio out Send 2, 3, or 4, respectively, and receive audio from Return 2, 3, or 4, respectively.

For stereo effects, with Helix Floor, FX Loop 1/2 sends audio to Send jacks 1 and 2. These patch to the stereo hardware's inputs. The stereo hardware's outputs re-enter Helix by patching to Return jacks 1 and 2. FX Loop 3/4 works similarly with Send jacks 3 and 4, and Return jacks 3 and 4.

With HX Stomp, FX Loop L/R's audio goes to the Send jack. To feed an effect with separate left and right inputs for stereo, you need a stereo-to-dual-mono breakout cable. The effect's stereo outputs then re-enter FX Stomp through Return L and Return R. Note that if the effect was mono in, stereo out (e.g., the late, great AdrenaLinn), you could use a standard mono cable with the Send jack.

The following parameters control FX Loop Send and Return characteristics:



- **Send** determines the level going to the external effect. If this parameter doesn't provide enough level, it may be necessary to change the Global setting for this loop to Line level. When running Helix as a guitar-level processor, the gain range will likely be sufficient.
- **Return** sets the level coming back into Helix from the external effects.
- **Mix** acts like a wet/dry control for external effects, so you can balance the level of the audio going into the effects loop with the audio coming out of it. Typically, you'd set this to 100% (wet sound only), unless you wanted to mix in some of the audio going to the external effect.
- **Trails** determines whether an external delay or reverb continues to decay after bypassing the FX Loop block or selecting a different snapshot. With Trails on, the decay continues. Trails off mutes the decay.

Send and Return Blocks

Send and *Return* blocks are like FX loops, but they allow sending audio from, and returning back into, different parts of the signal chain. For example, suppose you want a clean path through the Helix with some modulation, followed by reverb. But you also have an external hardware effects chain with a vintage fuzz and delay. You want to feed the external hardware with dry guitar, and then mix the hardware effect chain's dirty delay in parallel with the reverb.

Using an FX Loop block for the external chain would have to go either before or after the reverb—neither of which gives the intended result. Routing the signal through a separate Send and Return makes it easy to create this kind of parallel effect (fig. 2.110).

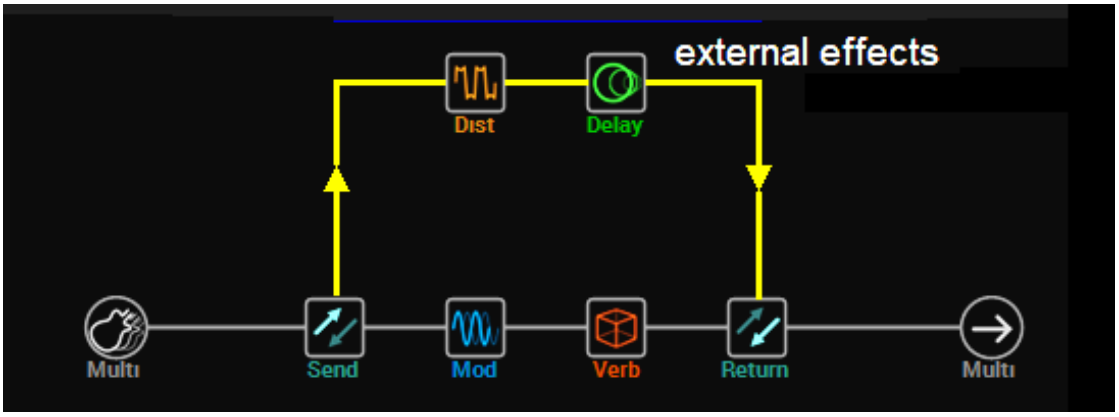
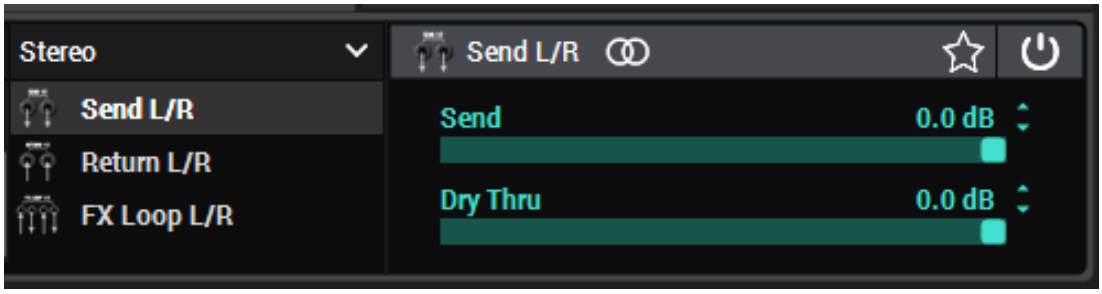


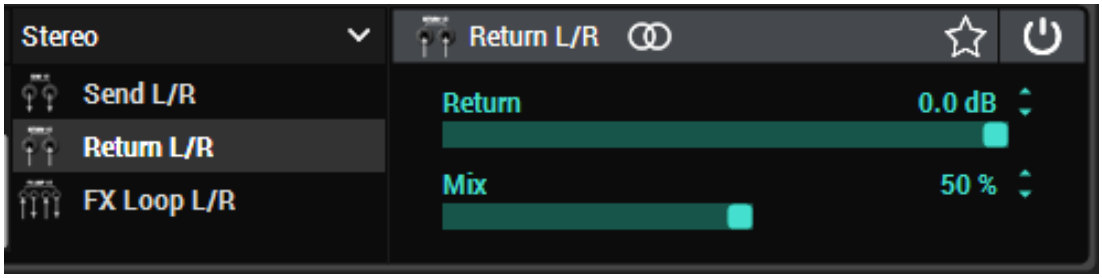
Figure 2.110 The upper blocks represent external effects. The yellow lines represent patch cords. The Send goes to the external hardware chain's input, whose output returns back into Helix via the Return.

The Send block parameters are:



- **Send** sets the output level that exits the Send jack to external hardware.
- **Dry Thru** adjusts the level of audio that passes through the Send block to subsequent Helix blocks. Send usually "picks off" a signal from Helix's signal path. Because the FX Loop is in parallel, turn Dry Thru up so you can hear the Helix path along with the parallel, external FX Loop.

The Return block parameters are:



- **Return** determines the level of the external hardware signal feeding into Helix.
- **Mix** sets the balance of the returned signal and the existing Helix signal path that Return rejoins. Because the FX Loop provides a parallel effect, this control will usually be around 50%.

Tip: Send almost always inserts before Return. However, Helix doesn't place any constraints on this—you can insert the Return *before* the Send, which is essential for using the 4-Cable Method (see next). But it also lets you create audio feedback loops, which have serious potential for sound design. Check out the tip Sci-Fi Feedback Effects Generator for a practical (albeit twisted) application.

Helix and the 4-Cable Method

Some physical amps have an effect loop, which adds a post-preamp send jack and a pre-power amp return jack. The preamp processes the sound, which then goes to the effects loop send. This patches to an external effect. The external effect's output returns to the guitar amp's return jack. So, if the preamp distorts the signal, you can place effects like reverb, chorus, and delay post-preamp distortion. This sounds cleaner than inserting these kinds of effects before preamp distortion.

The 4-Cable method allows placing some Helix effects in front of a guitar amp's preamp, and other Helix effects in the amp's FX Loop (fig. 2.111). This requires using Helix's Send and Return blocks.

Let's do a step-by-step analysis of the routing:

- The Helix Send patches to the physical amp's input. The effects between the guitar and the Send block go directly into the guitar amp's input.
- The large "X" in fig. 2.110 represents that we don't want the initial Helix signal path to go past the Send block. So, set the Send's Dry Thru to -120.0 dB.
- The physical amp's FX Loop Send jack patches to the Helix Return jack. We want only the sound from the amp's FX Loop, so the Return 1 Mix setting is 0% (i.e., only the Return signal goes into Helix).
- The effects loop goes through two Helix effects (in this example, Delay and Reverb). Helix's output connects into the amp's FX Loop Return jack, and from there, to the guitar amp's power amp.

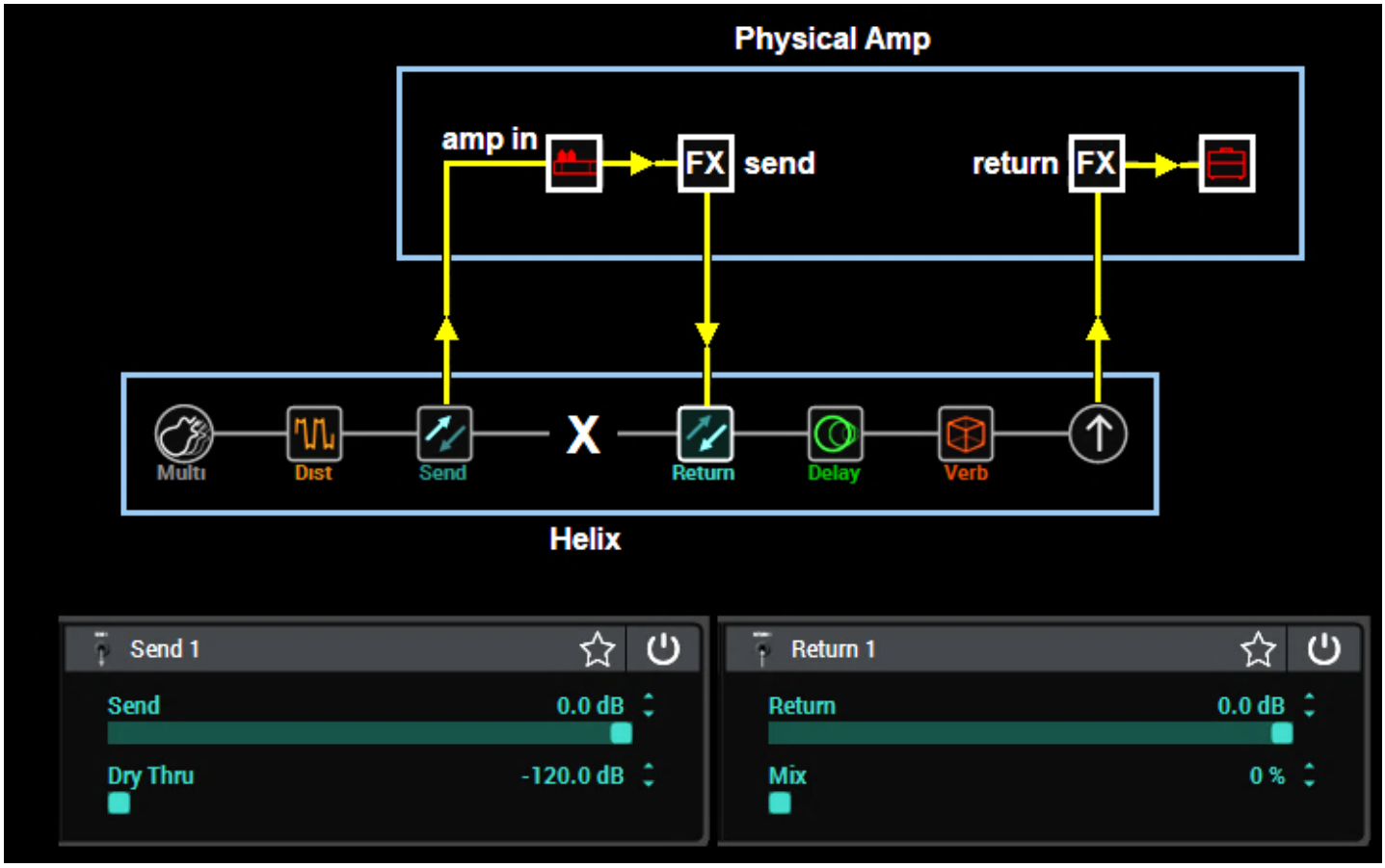


Figure 2.111 4-cable Helix setup, with the parameters for the Helix Send and Return blocks.

Fig. 2.112 shows a variation on the above. There's a trend for players to recognize the value of in-ear monitors (IEMs) to protect their hearing from high stage volume levels. However, sometimes it can be difficult to generate a signal from your amp to feed into the IEM for monitoring. Some musicians take out one ear's IEM so they can hear the sound from their amp. Unfortunately, that defeats the IEM's purpose. Doing so puts your hearing at risk.

This application starts with the standard 4-Cable Method described above. However, then it adds an amp and cab that's as close as possible to what you use onstage. This lets you mix in amp sound into the personal mixer that carries your IEM feed. Although the sound is not exactly what you'd hear from your amp, Helix's selection of amps is varied enough to provide a similar, if not almost identical, sound.

Note that in this setup, the Helix Send that connects to the physical amp's FX Loop return would need Dry Thru set to 0.0 dB. Then, the signal can pass through to the final Helix Amp+Cab stage.

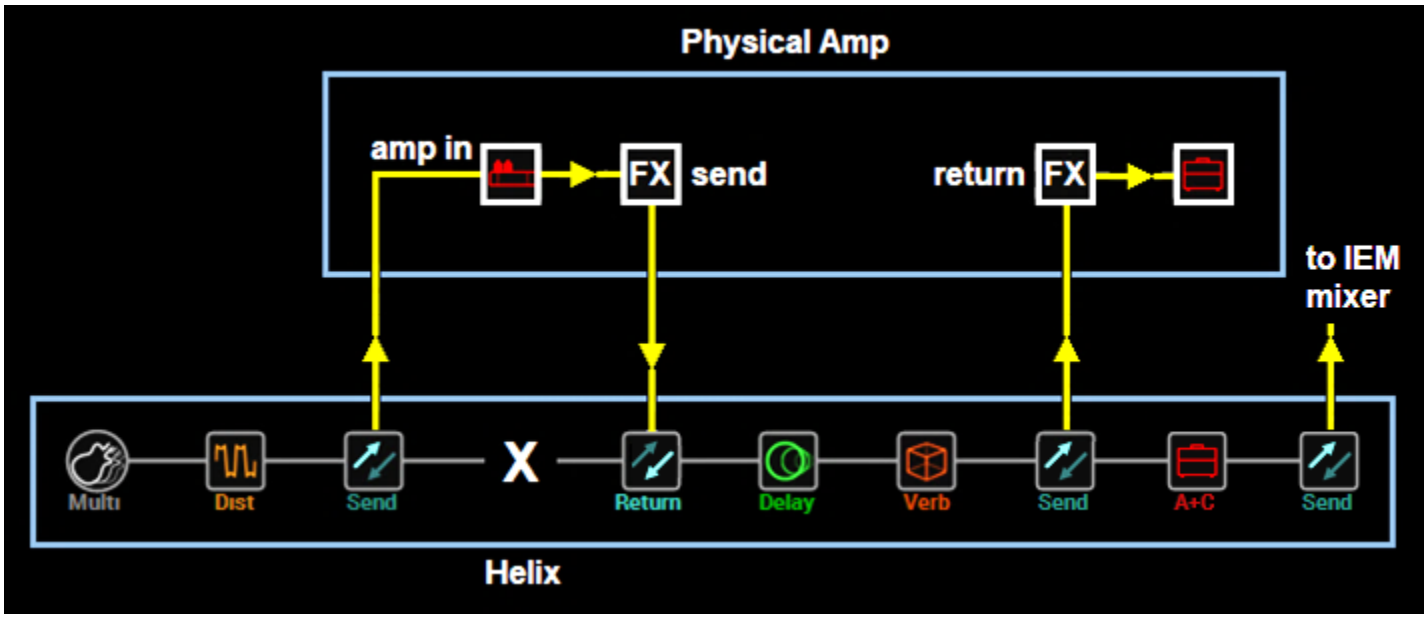


Figure 2.112 Workaround for mixing an amp sound into your IEM monitor feed. Use this when it's not practical to obtain a feed from the physical amp itself.

Amp Blocks

Most users are familiar with parameters like Bass, Treble, Bright, etc. We'll concentrate on some of the more unusual parameters that are common to various amps. Then, we'll do a deeper dive into amps that were introduced in firmware updates 3.15, 3.60, 3.70, and 3.80.

Common Amp Parameters

Let's start with parameters some Helix users find perplexing.

Hum

This adds a controllable amount of hum, to give a "you're listening to a real amp" feel. The left half of fig. 2.113 shows the hum at maximum. The right half shows a single note hit with moderate force.

Figure 2.113 Hum (left) and guitar note (right).

How to Hear the Way Hum Affects the Sound

1. Load the WhoWatt 100 block with no cab. Set Drive to 0.0.
2. Turn Ripple all the way down.
3. Turn up Hum to 10.0
4. Hum produces a low-level, low-frequency signal. You'll probably need to turn up your monitoring system's level to hear it.

Ripple

Ripple in physical amps is due to power supply-related issues. In theory, power supplies filter the incoming AC (alternating current) power into DC power, but it's almost impossible to avoid some residual AC ripple. Because the power supply affects every part of an amp, even subtle imperfections compared to pure DC power may affect the sound. Ripple can be more pronounced when an amplifier is under heavy load, or when the filter capacitors in older amps have aged.

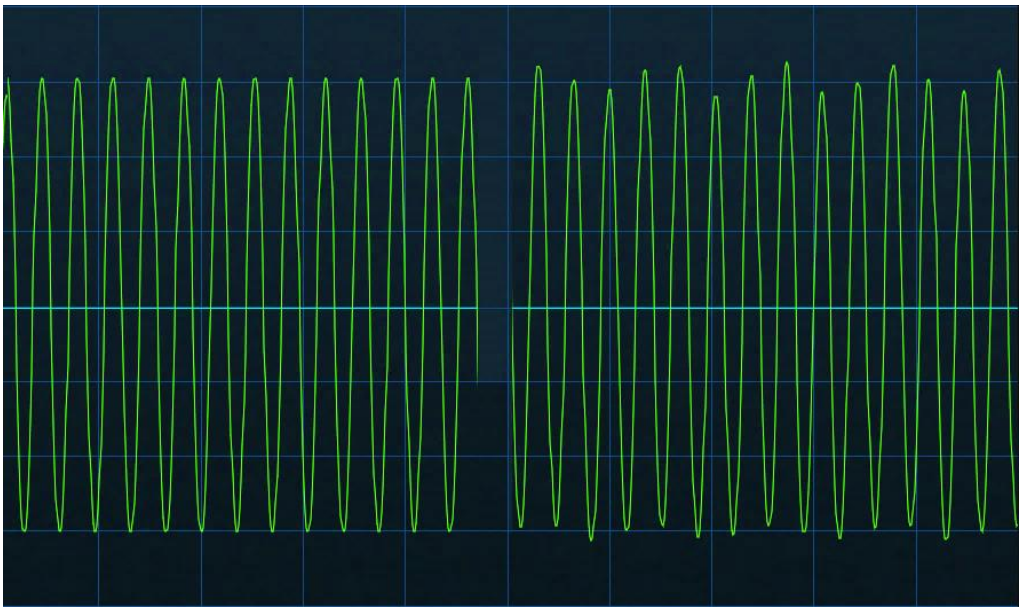


Figure 2.114 A sine wave (left) being affected by ripple at maximum (right).

In fig. 2.114, the input signal is a 400 Hz sine wave. Unlike hum, which is in parallel with the audio, ripple modulates the audio level. This effect is audible, and measurable, even with moderate amounts of distortion.

How to Hear the Way Ripple Affects the Sound

1. Load the WhoWatt 100 block with no cab. Set Drive to 0.0.
2. Turn Hum and Ripple all the way down.
3. Turn up Ripple to 10.0. At a sufficiently loud monitoring level, you'll hear something that sounds like hum, but with a more complex waveform.
4. Play a note above the 12th fret on your high E string. With ripple up all the way, you'll hear a slightly grittier sound "underneath" the note. Turn ripple down, and the sound will be cleaner.

Ripple is more of a subliminal effect. Without ripple, notes sound cleaner. However, some guitarists feel that adding ripple makes amps sound more alive and interesting, because ripple creates complex overtones. Try boosting ripple on some favorite presets and play through them for a while. Then turn ripple all the way down and play through the same presets some more. Choose whichever option puts a smile on your face.

Bias and Bias X

There's no universal "rule" on how to adjust these parameters. Bias and Bias X interact with each other as well as various other settings—the input signal level, drive amount, sag, and master volume. So, a Bias setting that increases 2nd harmonic distortion may increase 3rd harmonic distortion with other control settings.

However, let's show how varying these controls affects the way distortion emphasizes different harmonics. Fig. 2.115 shows the WhoWatt 100 with both Bias X and Drive at 5.0. The top image shows Bias at 0.0. The bottom image shows Bias at 10.0 The even harmonics have yellow circles. The odd harmonics have red rectangles.

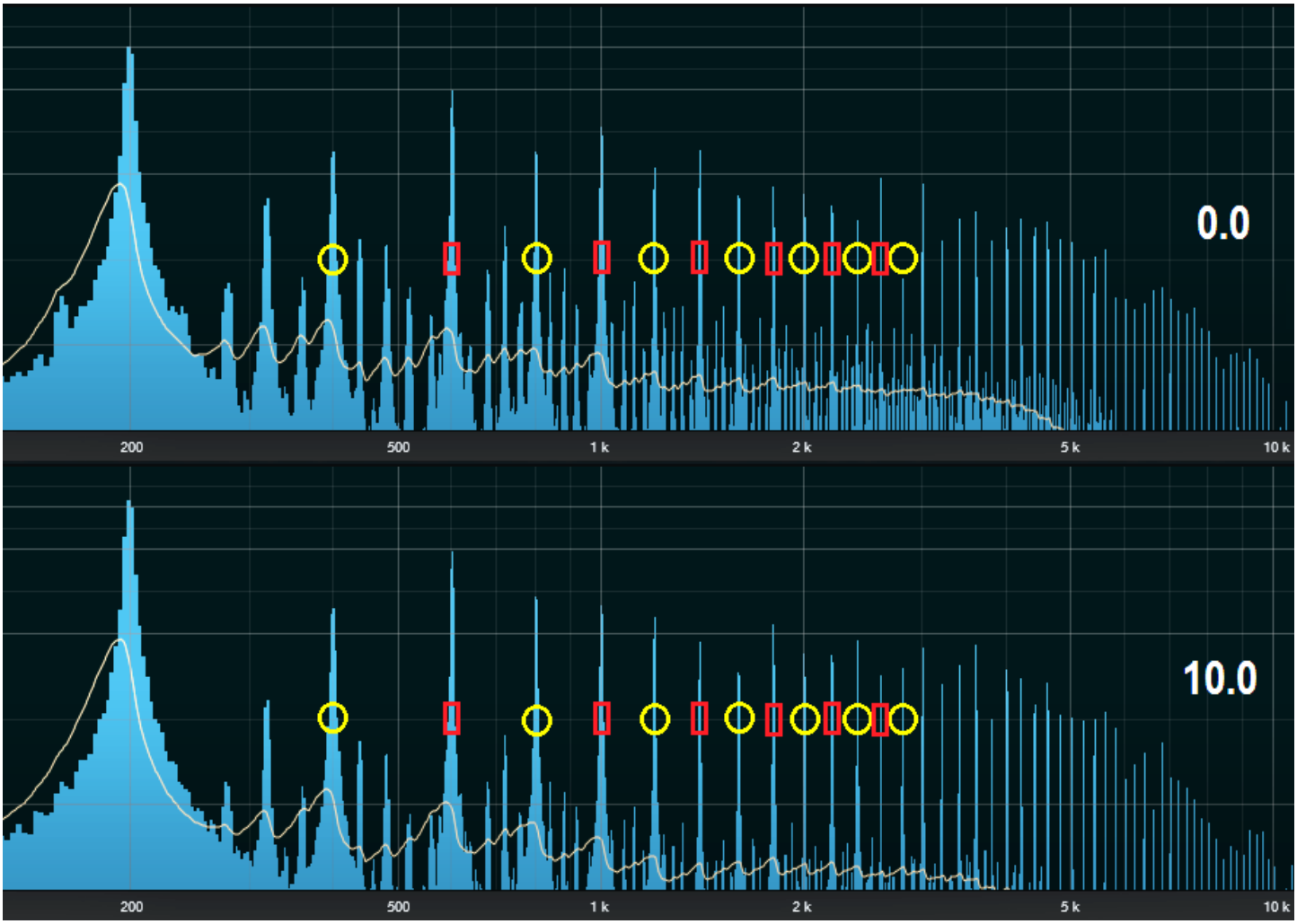


Figure 2.115 Odd- and even-harmonic distortion for two different Bias settings.

While it's not an exact science, the 10.0 setting is more prone to emphasize even harmonics, especially at higher frequencies. It also generates less overall distortion. 0.0 is more likely to emphasize odd harmonics. But again, remember that changing any of several parameters could give different results.

Tech Talk: About Bias X

Ben Adrian, Line 6 Sound Design Manager, explains it as follows: "All tube amps need to bias the power amp tubes, usually by applying a negative voltage to the input audio signal. However, when (and only when) the power tubes are distorting, free electrons can form around the input grid and shift in the bias voltage. This shift causes a tonal and texture change much like adjusting the bias control. However, once the tubes leave the distorted state, the free electrons dissipate and the bias returns to normal. Another way to put it (less accurately) is that this is a level/envelope controlled bias shift. It's a very subtle change, so please don't expect high drama from this parameter."

How to Hear the Way Bias and Bias X Affect the Sound

Ultimately, what matters is how your guitar sounds with different settings. However, sending a sine wave through an amp, and listening to how Bias and Bias X affect the sound, reveals a lot. Fortunately, Line 6 has the ideal tone generator. Try the following (fig. 2.116):

1. Insert the 4 OSC Generator in front of the Amp+Cab you want to test, or at the beginning of a preset that you want to edit.
2. Turn down all 4 OSC Generator level controls except for Osc 1 Level, and the main Level (fig. 2.115). A 200 Hz sine wave is a suitable test tone.

Experiment with different Bias, Bias X, Drive, Sag, Master Volume, and other parameters. As you change the Bias and Bias X sliders, you'll hear different distortion characters. Also, listen to low Drive settings as you change the bias. You'll find almost all the amps offer useful variations just by altering the Drive, Bias, and Bias X parameters.

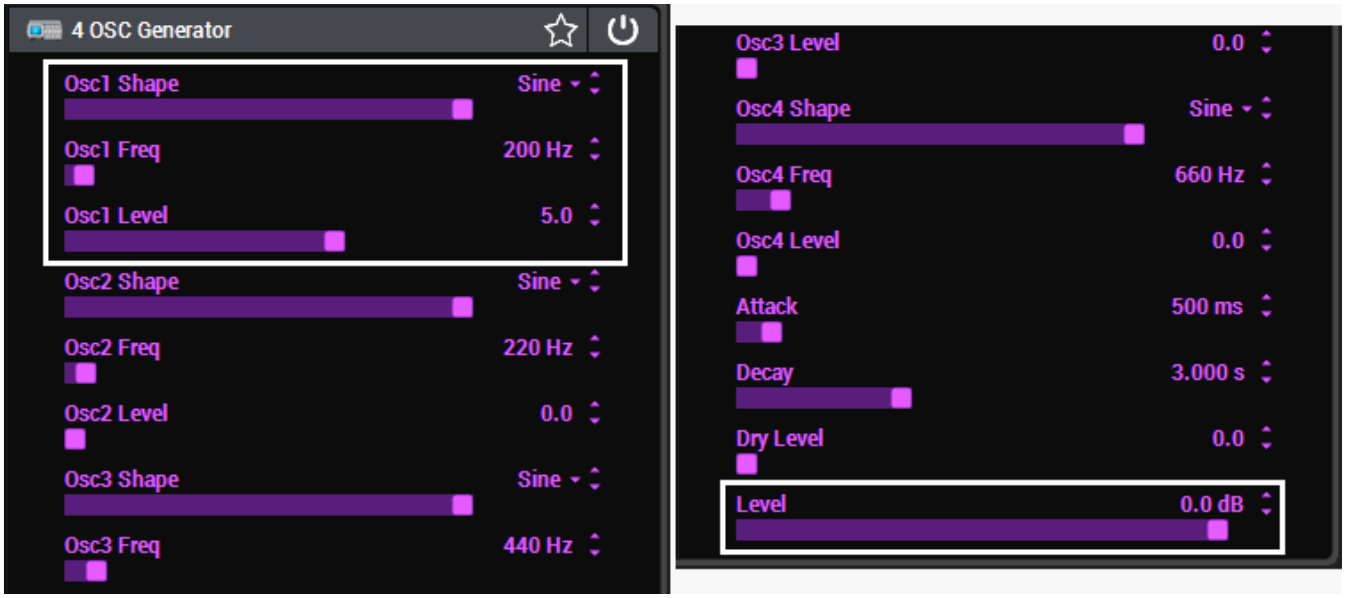


Figure 2.116 With Osc2, Osc3, and Osc4 Levels set to 0.0, the only controls that matter are the ones with white outlines.

Sag

An amplifier doesn't necessarily react predictably when you hit it with a note or chord. The sudden demand for current from the power supply can cause the voltage to start off higher, sag, and then stabilize as the amp delivers more current. Or, it may sag at first, and then recover. How the output transformer reacts also matters.

Fig. 2.117 shows several amps' sag characteristics, with Sag set to maximum. The Tweed Blues Nrm has the most anomalous sag, followed by the Brit Plex Nrm and WhoWatt 100. The Essex A30 has a more controlled sag, as do the Grammatico Norm and Das Benzin Mega. The time scale for all amps is the same. Sag occurs over a short amount of time.

With Sag at minimum, the output level doesn't change when a strong input signal hits the amp. This diagram also shows why you can't make an overall conclusion about sag, like "more sag gives more sustain" or "more sag gives more touch sensitivity." These comments are true for some amps, but not for others. Also remember that sag interacts with other parameters. The one comment that *does* apply is that setting Sag to minimum gives a "tighter" sound, because the attack is instantaneous.

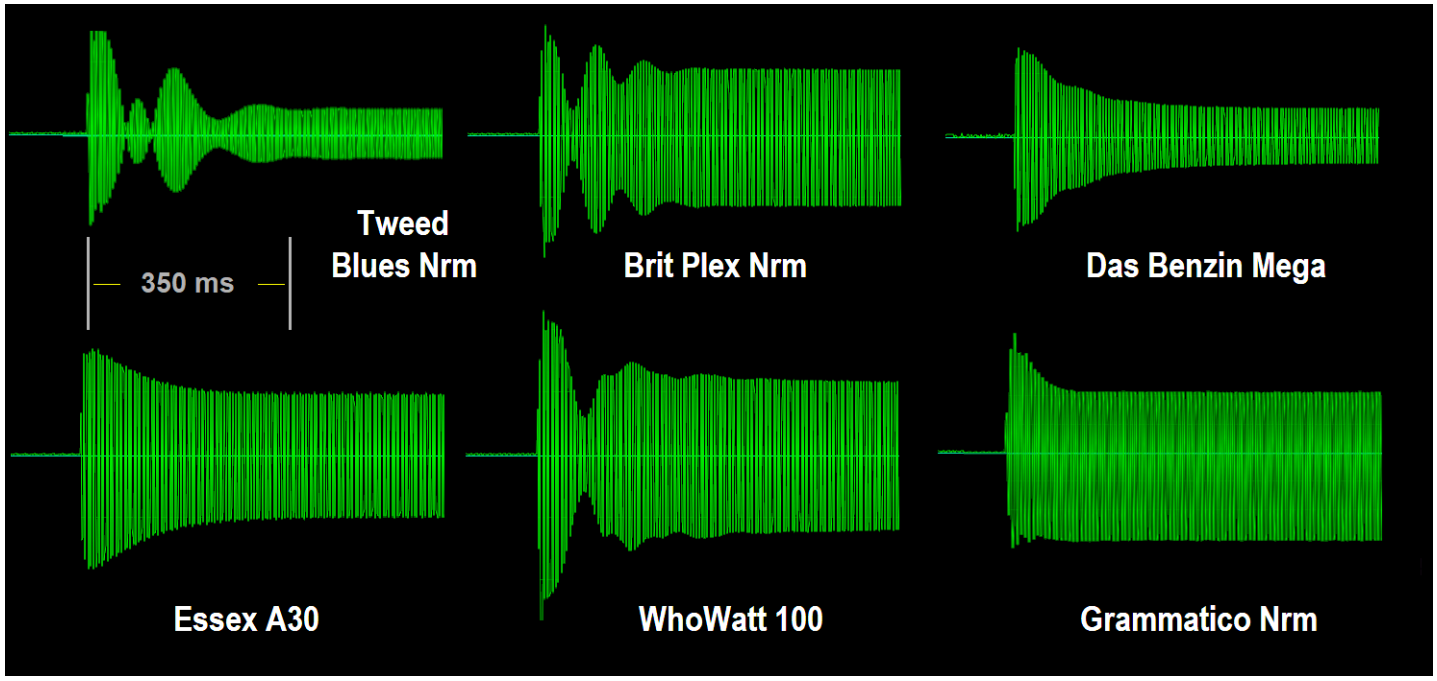


Figure 2.117 Sag-influenced attack characteristics for six different amps.

How to Hear the Way Sag Affects the Sound

As with testing Bias, we'll take advantage of the 4 OSC Generator. Here's how to run the test:

1. Insert the 4 OSC Generator in front of the Amp+Cab you want to test, or at the beginning of a preset that you want to optimize.
2. Use the same settings as in fig. 2.116 but set Attack and Decay to minimum.
3. Turn on the 4 OSC Generator while listening to the amp output.
4. To evaluate different sag settings, bypass the 4 OSC Generator block, change the Sag control, and then enable the 4 OSC Generator block again.

You can't lower the 4 OSC Generator's attack time below 10 ms, so ideally, you'd use a test tone generator with an instant attack time. Your DAW may include a test tone generator. You'll also find third-party options like the Waves eMo Generator plug-in.

Ch Vol

This controls the overall output level. Varying it doesn't affect tone or distortion.

Line 6 Ventoux Amp

This Line 6 Original Amp is different enough from other amps that we'll do some analysis to better understand it. Sag, Hum, Ripple, Bias, and Bias X work the same way as other amps. However, the "tone control" stack takes a new approach. In all the tone control screenshots, there's no cab, only the Ventoux amp. So, the frequency response changes relate *only* to the amp. First, let's look at the amp's sag (fig. 2.118).

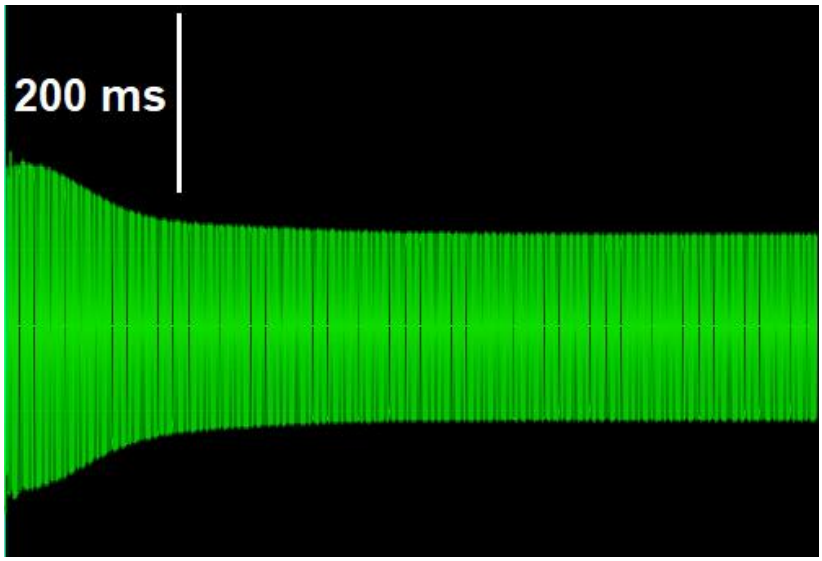


Figure 2.118 Ventoux amp sag at 10.0.

This is a well-behaved sag, not unlike the Essex A30. There's no sag with Sag = 0.0.

Depth

Depth is like a Bass control. However, it's more about adding resonance than conventional shelving- or peak-type bass controls. Fig. 2.119 was taken with HP, Mid, and Presence at 5.0.

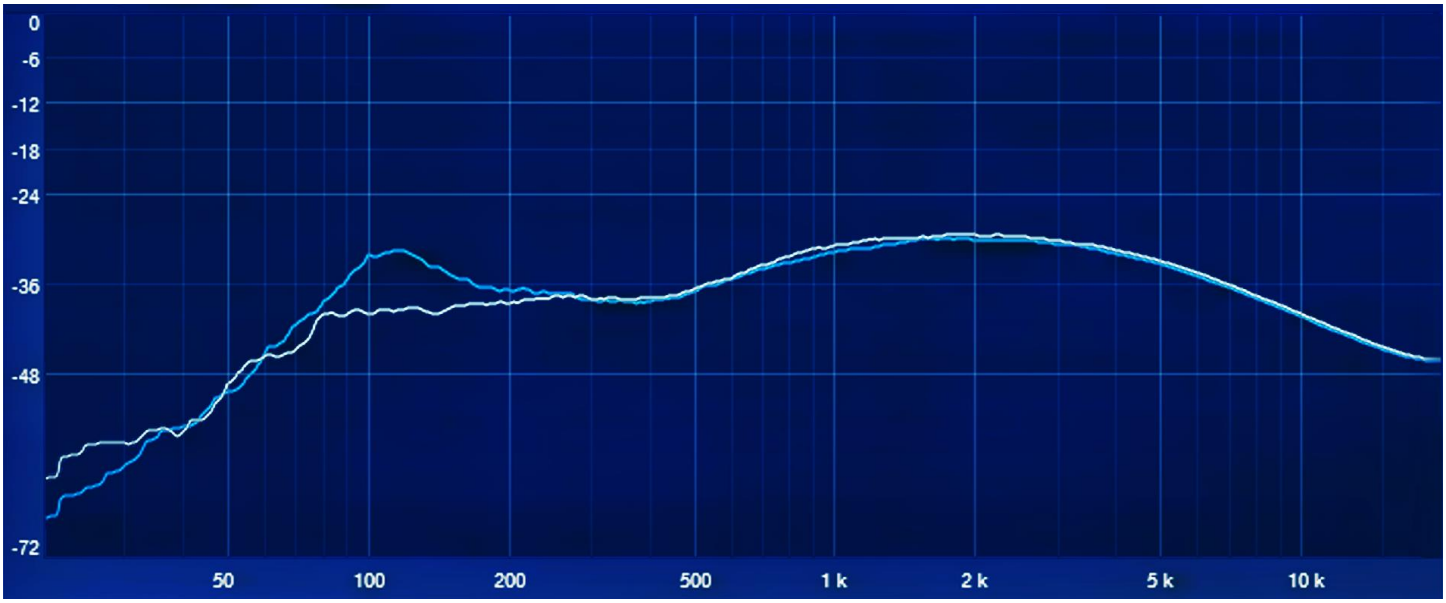


Figure 2.119 The white line is Depth = 0, the Blue line is Depth = 10.

HP Filter

HP (High Pass) Filter is an unusual type of tone control. Turning it up reduces the low-frequency response but also adds a little boost at the highs. This combination thins the sound. Turning HP Filter down restores the original bass response, which gives a warmer tone. In fig. 2.120, Mid and Presence are at 5.0, Depth at 0.0.

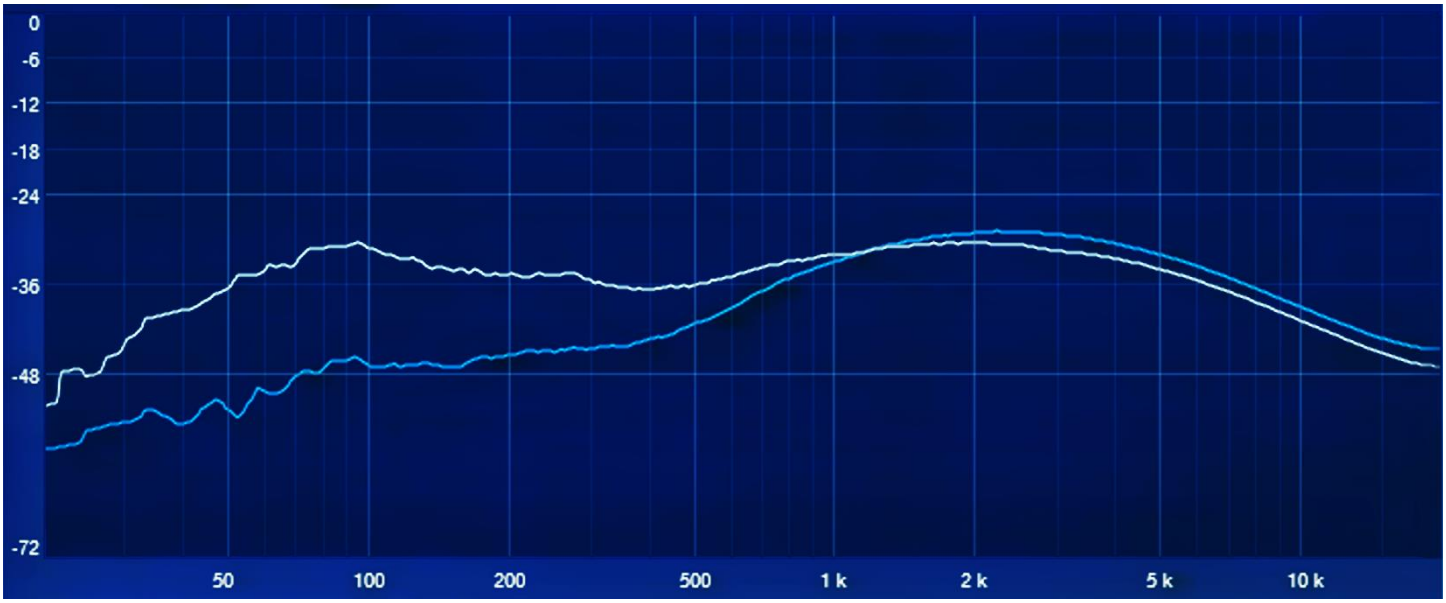


Figure 2.120 The white line is HP Filter = 0, the blue line is HP Filter = 10. Note the reduced low-frequency response with HP Filter turned up, but there's also a subtle high-frequency boost.

Presence

Presence emphasizes the highs, starting in the midrange. It also de-emphasizes the lower midrange frequencies. This reduces the potential for a "muddy" sound. However, this reduced frequency response in the lower mids implies that if you want more treble, you'd probably want to use Presence in conjunction with the other controls (like turning up HP Filter). In fig. 2.121, HP Filter and Mid are at 5.0, Depth at 0.0.

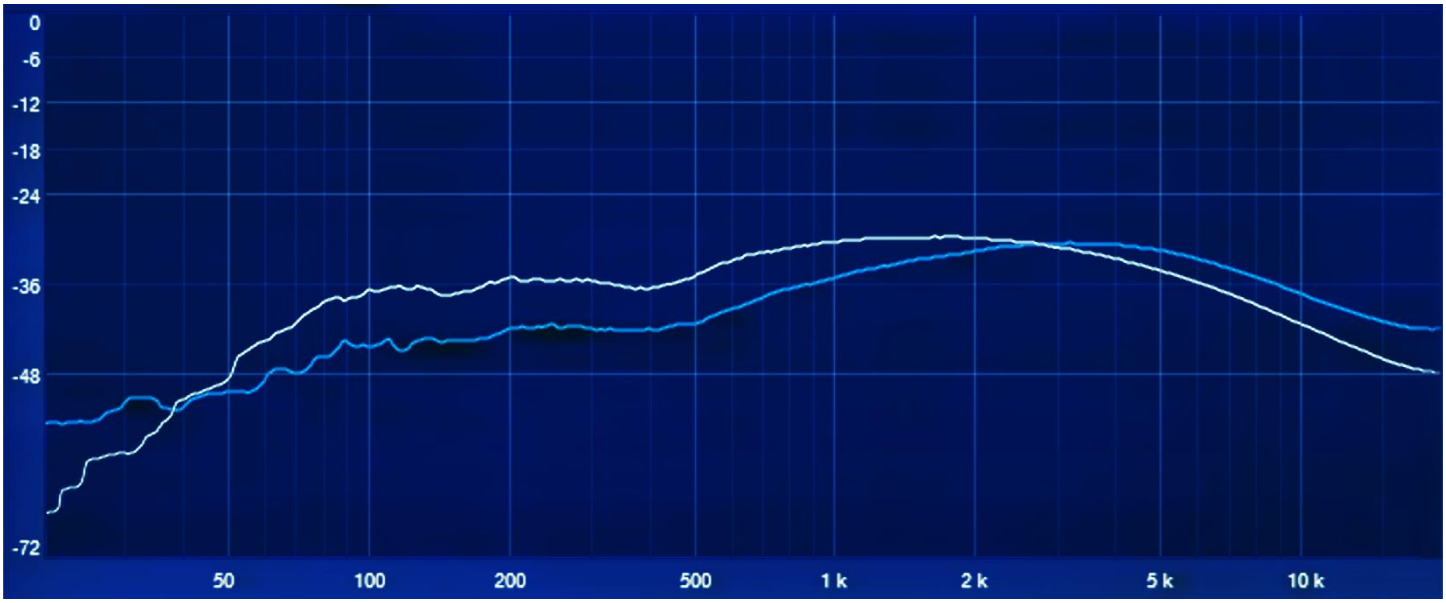


Figure 2.121 The white line is Presence = 0, the blue line is Presence = 10.

Mid

Mid increases the response from about 200 Hz to 1 kHz. In fig. 2.122, HP Filter and Presence are at 5.0, Depth at 0.0.

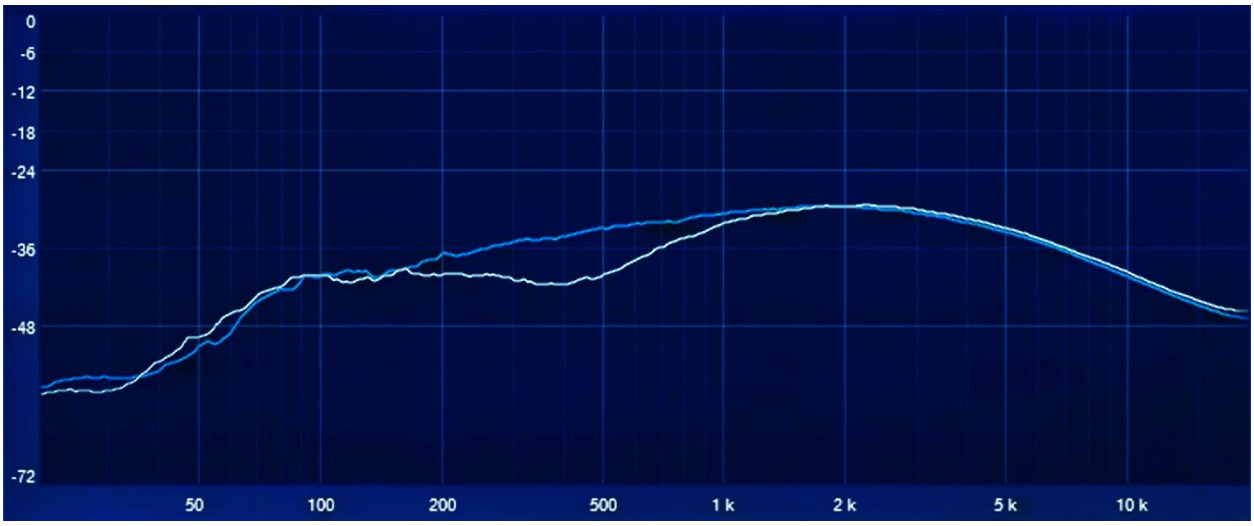


Figure 2.122 The white line is Mid = 0, the blue line is Mid = 10. Try various Mid settings when the amp is crunching.

When Worlds Collide

Sorry, I couldn't resist the silly heading. But because the parameters interact, changing several of them can have a major effect on the overall sound. Fig. 2.123 shows two examples. The white line is almost a flat response. The parameter settings are HP Filter, Bass, and Presence at 0.0, with Mid = 10.0 The blue line shows a scooped response. Control settings are HP Filter = 6.0, Mid and Presence = 0.0, and Depth = 10.0.

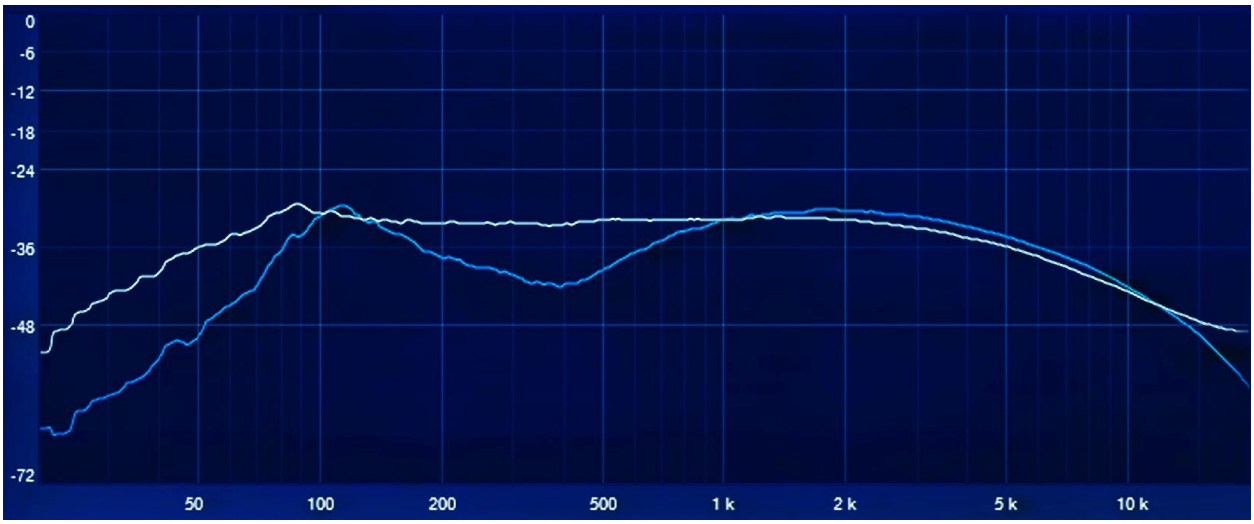


Figure 2.123 Interacting controls produce multiple potential equalization curves.

Master Volume

The Master volume parameter is more flexible than some amps. It's intended to be left at 10.0, which recalls vintage amps with no master volume. But this underscores the value of virtual amps, because reducing this level produces a variety of less dramatic, but useful, tonalities.

Grammatico GSG

The Grammatico GSG100 has features and controls inspired by the Dumble Overdrive Special. The Overdrive circuitry added two tube gain stages after the traditional tube stages used for clean tones. This provided a creative alternative to the lead channels in other late 70s/early 80s guitar amps. As a result, the Drive, OD Drive, and Master volume controls interact. This can lead to great sounds but may make dialing in the desired tone a challenge. There are so many possibilities that altering even a few parameter settings can make the difference between ugly and beautiful sounds.

This section combines my observations with the release notes, which are quite detailed. To use this amp to its fullest, experiment with the various parameters to hear how each one affects the sound.

Drive

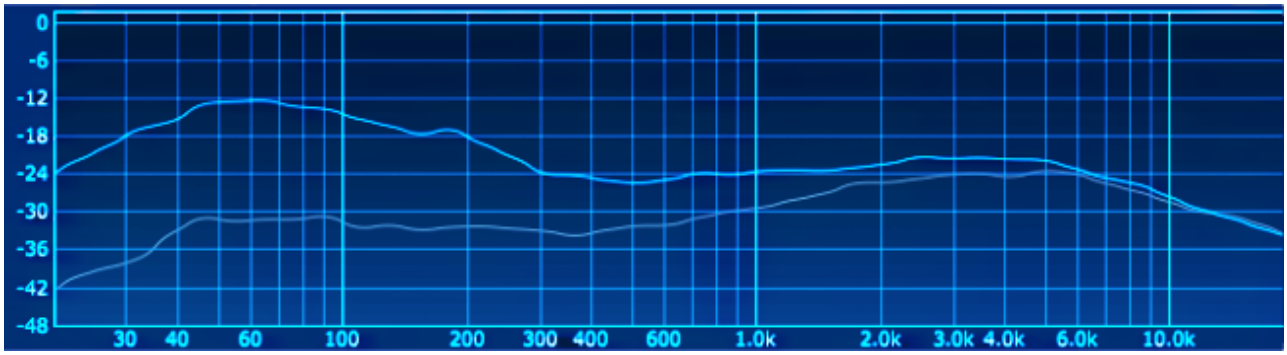
This is the amp's first volume control. On the original amp, this control was called "Volume." Drive maintains naming consistency with other Helix models.

Bass, Mid, and Treble Tone Controls

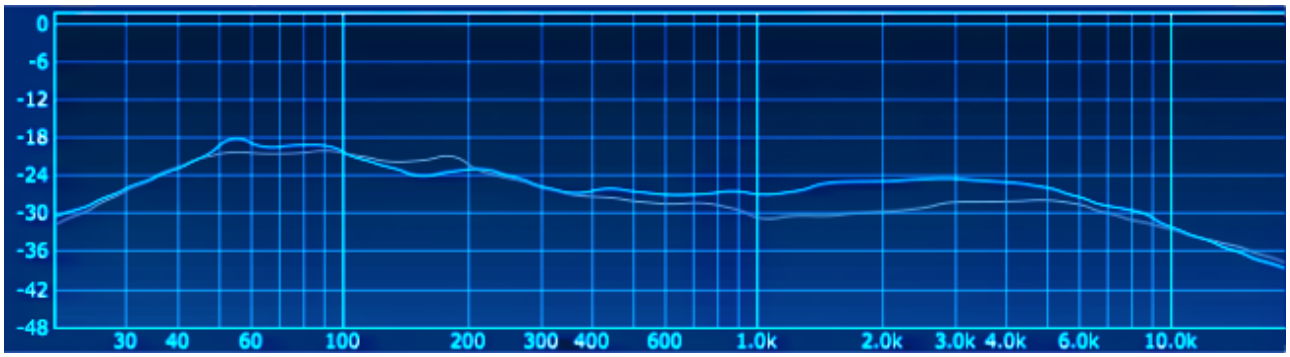
These controls model passive EQ inserted between the preamp's first and second gain stages. Their ranges differ from traditional amp tone controls. An additional Rock/Jazz switch changes the tone stack's voicing.

The following curves show the Rock and Jazz settings. The brighter curve is the control setting at 10.0. The lighter curve is the 0.0 setting. The other tone knobs are in their default positions. The Mid switch is off.

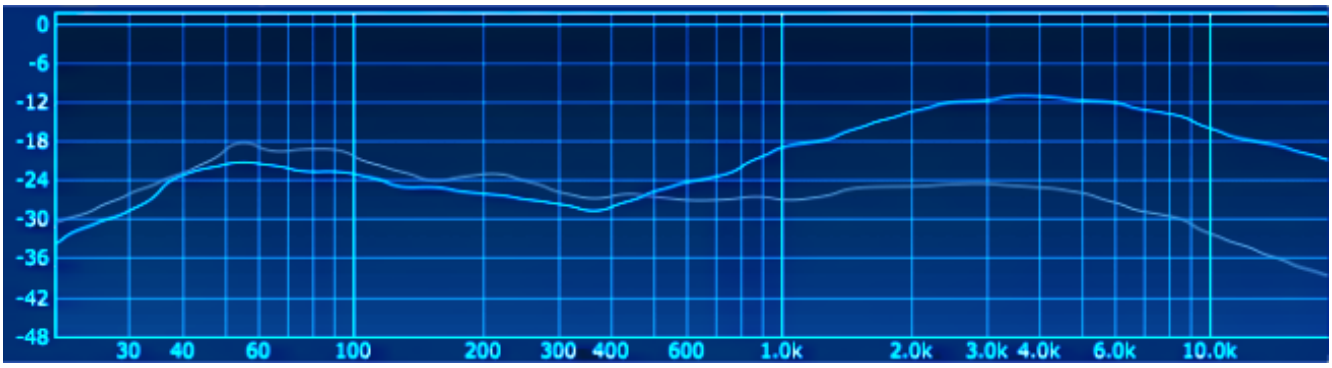
Rock Bass



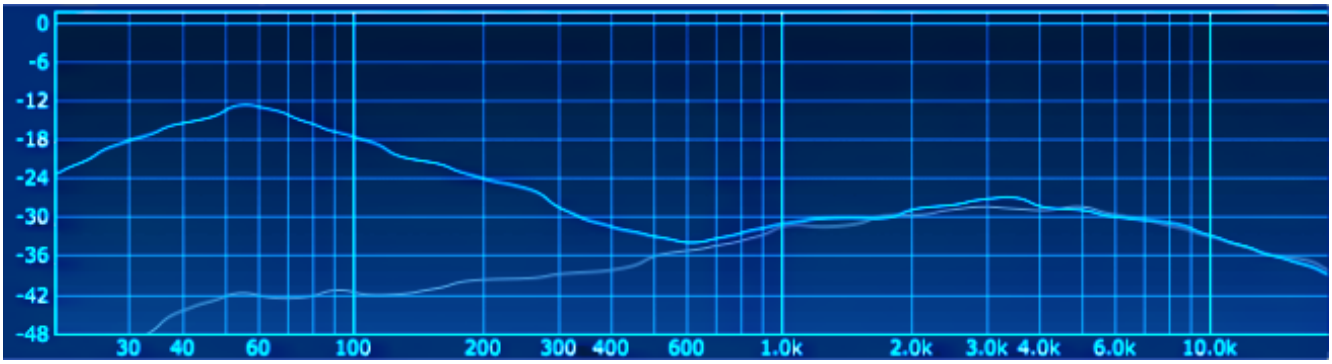
Rock Mid



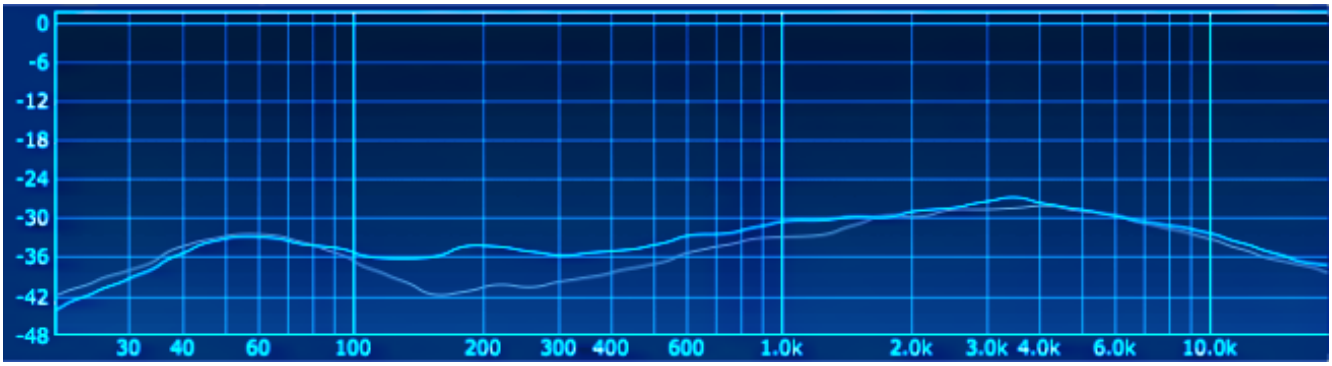
Rock Treble



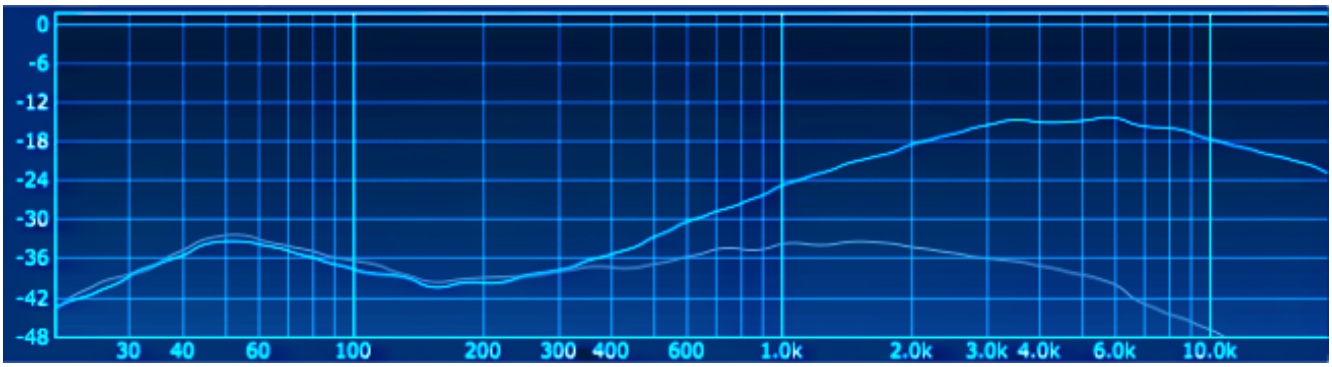
Jazz Bass



Jazz Mid



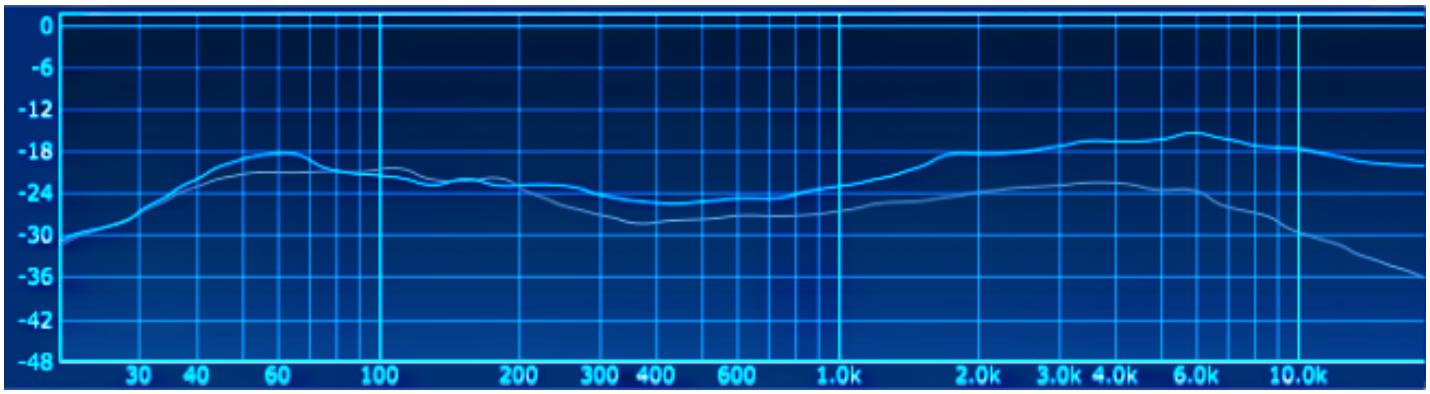
Jazz Treble



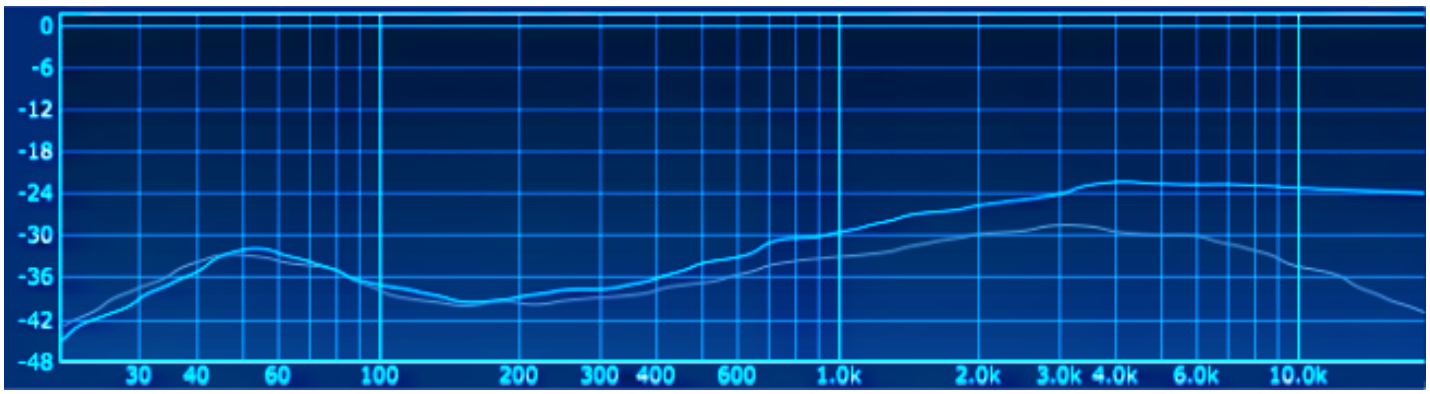
Presence

Presence is like the presence control in other guitar amps. It changes the power amp's high-frequency response by modifying the EQ filtering in the power amp's negative feedback loop.

Rock Presence



Jazz Presence



Ch Vol

This controls the amp's output level. It has no effect on tone or distortion characteristics.

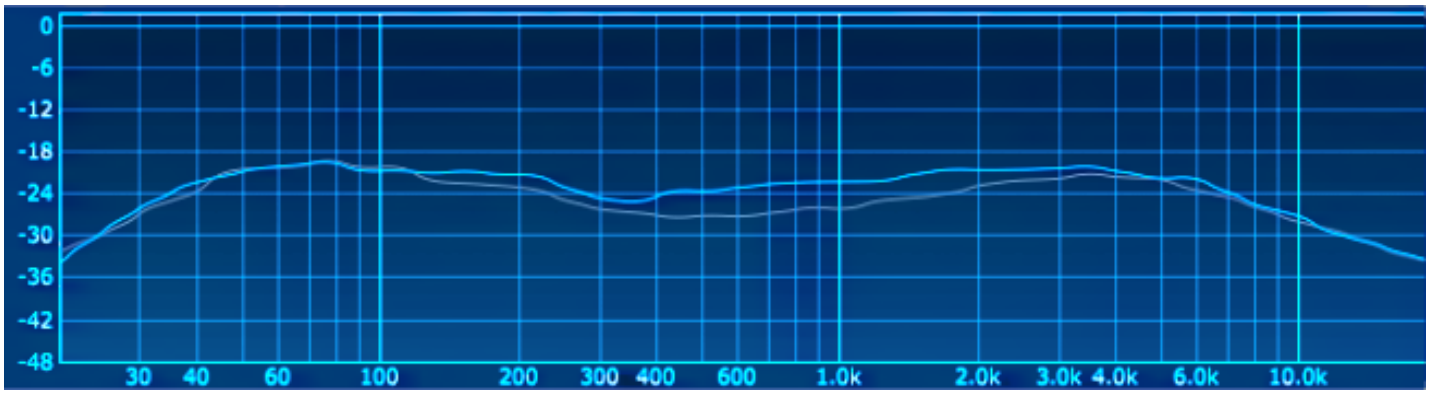
Master

The master volume inserts between the preamp and power amp, so it can dial in more or less power amp distortion. This amp is *very* loud, and most players would probably never crank the master volume in real life. It can sound nasty, and not necessarily in a good way. Keeping it on the low side usually sounds best.

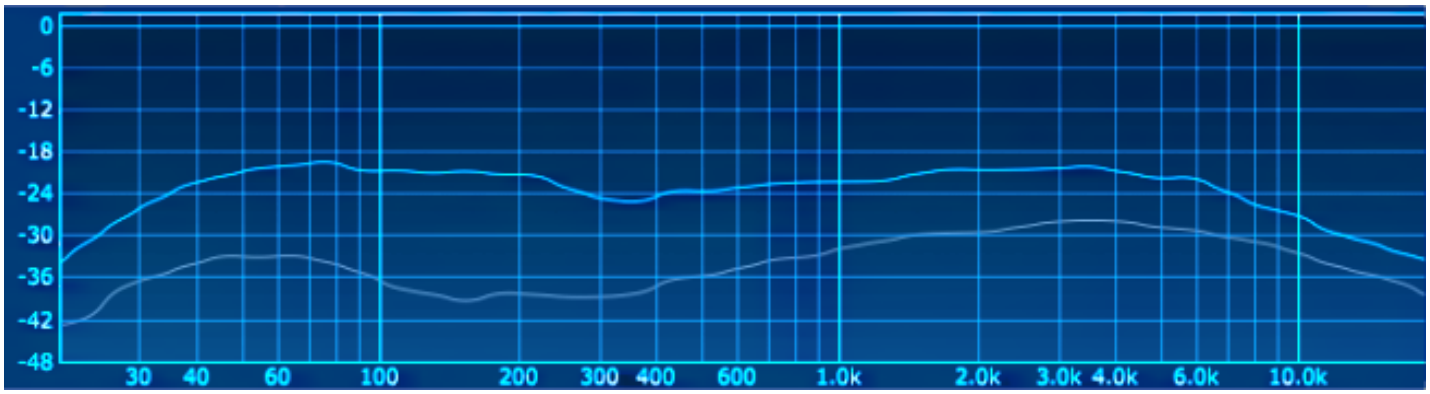
Mid Switch

This switch changes the value of the tone stack's treble capacitor. The Off setting gives the amp a more scooped sound. Enabling Mid adds a noticeable upper-mid boost. Note that the Off curve is the same as the default tone control settings.

Rock Mid Switch



Jazz Mid Switch



Jazz/Rock Switch

This switch provides two separate tonal voices by changing the tone stack's wiring. Jazz is quieter, with a lower center frequency for the mids. Rock is louder. It has a more traditional mid-frequency center.

Tone controls rarely translate well between the Jazz and Rock settings. Settings that sound good in one mode don't necessarily sound good in the other mode.

OD Switch

This turns the two-gain-stage tube overdrive circuit on and off. Because the overdrive circuit is *after* the tone controls and Drive knob, turning on overdrive is like adding a third and fourth gain stage to the preamp. Line 6 recommends initially setting up the amp's base tone with the Drive and tone controls. Then, engage the OD switch. Finally, adjust the overdrive circuit to enhance the base tone.

OD Drive

This alters the overdrive circuit's amount of saturation. Because the overdrive circuit is after the amp's regular Drive and tone controls, the OD Drive knob's range depends on those earlier knob settings.

OD Level

Vary the overdrive circuit's output level with this control.

Bright

This three-position switch chooses between off, and one of two different bright capacitors. These are part of the Drive knob circuitry. The Bright switch's effect is most pronounced with lower Drive settings. It becomes less effective at higher Drive settings. With Drive set to 10.0, changing the Bright setting has no effect.

FET Boost

The GSG100 has a solid state, FET (Field Effect Transistor) boost circuit. It provides the same function as placing a FET Boost pedal before the amp. The physical amp has two input jacks, but the Helix model uses a switch to enable or bypass boost. This has the benefit of making the FET boost circuit controllable via footswitch. Turning FET Boost on also changes the EQ slightly.

PAB

PAB stands for "Preamp Boost," which removes the tone controls from the circuit. This restores any signal level lost due to the tone controls between tube gain stages 1 and 2. The tradeoff is that the tone controls no longer affect the sound, with two exceptions:

- The treble knob has a very minor effect.
- The mid switch changes the amount of lows and mids going through the circuit.

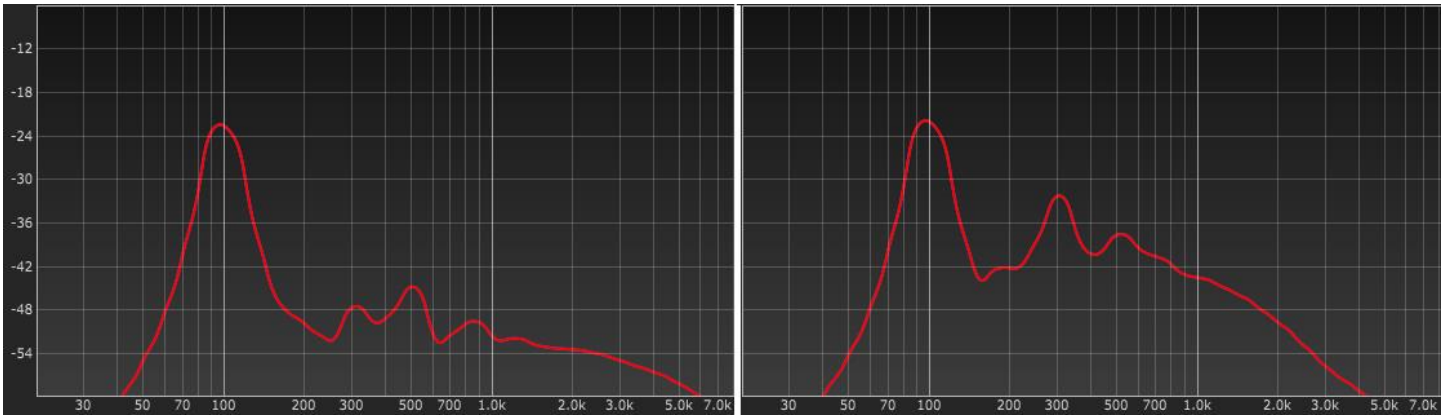
Other Controls

Sag, **Hum**, **Ripple**, **Bias**, **Bias X**, and **Ch Vol** are covered in the section on Common Amp Parameters.

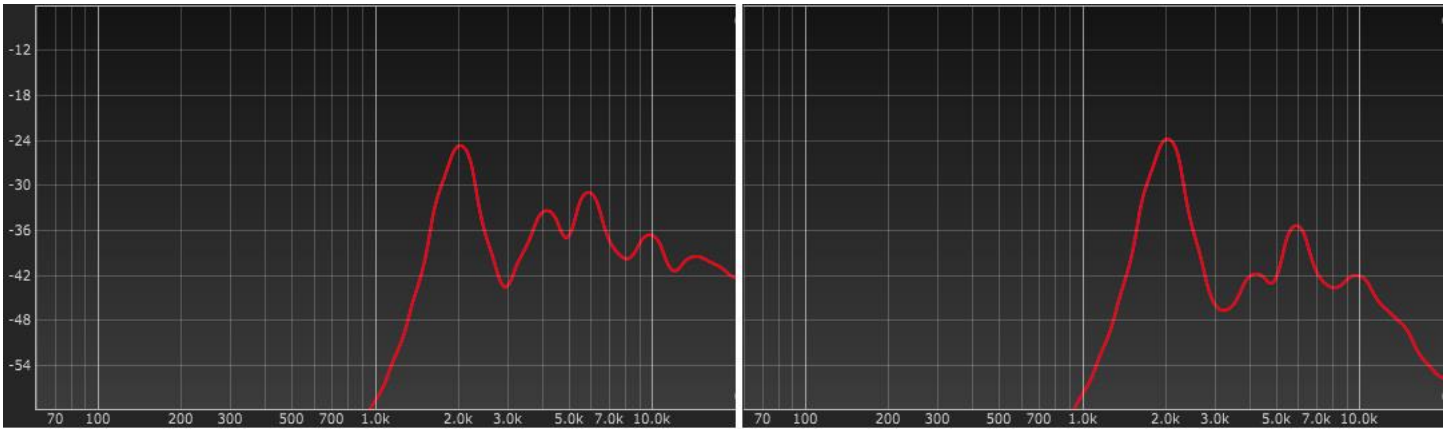
Line 6 Elmsley

The Elmsley is a Line 6 original amplifier that layers parallel distortion across the lower and upper registers. *Personal bias alert:* I like that Line 6 is getting more into designing original amps and effects. They're some of my favorite models.

The low end sounds full, and the upper mids and highs have excellent definition. If you're curious about why, check out the following graphs. The Elmsley response is on the left, and the Moo)))n Nrm on the right. Both amps use their default settings, without any cab.



With a 100 Hz sine wave, the Elmsley doesn't spill over much into higher frequencies. This underscores how the parallel distortion can emphasize the lower and higher registers independently.



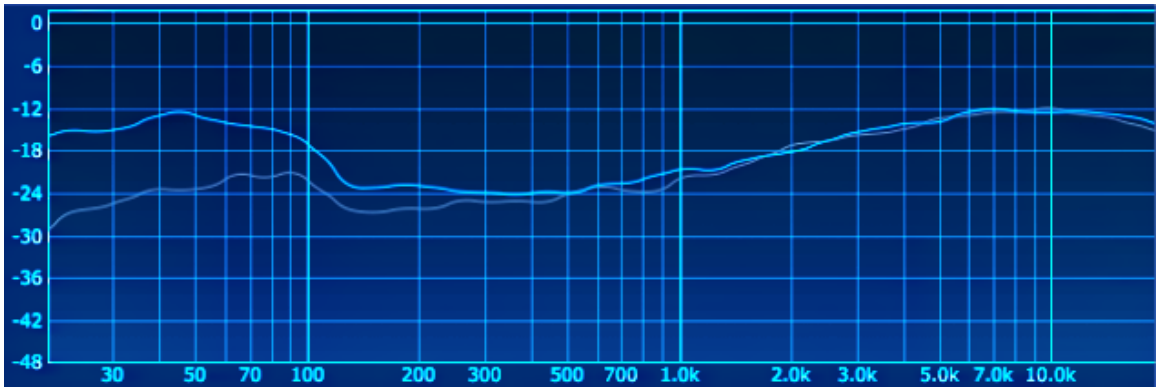
This shows the response with a 2 kHz sine wave. The Elmsley maintains its definition in the high end, which is why the amp has sparkle and definition.

In the tone control frequency response curves below, the brighter curve is the control setting at 10.0. The lighter curve is the 0.0 setting. Tone knobs other than the one being measured are in their default positions.

Drive

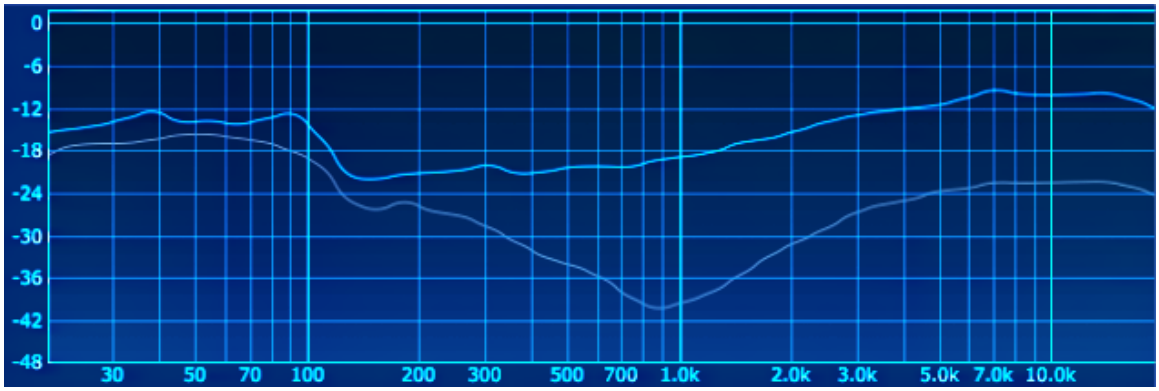
Drive controls the preamp level. This influences the amount of distortion.

Bass



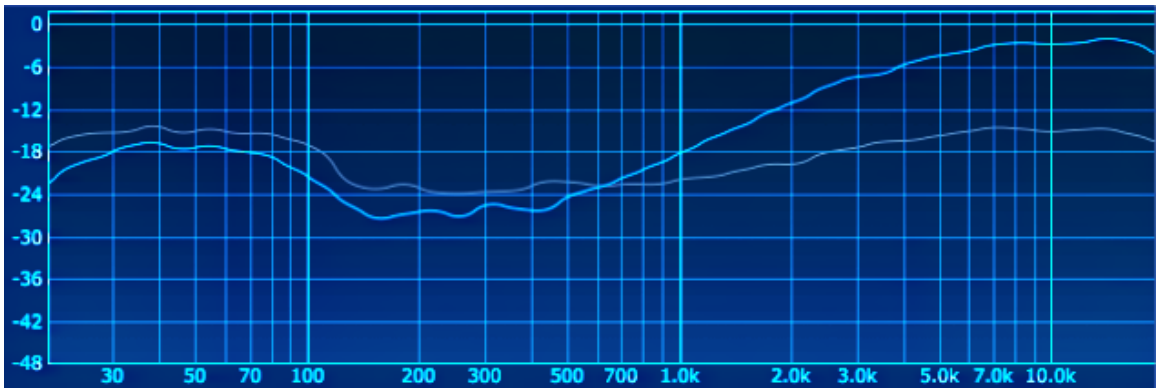
In all the frequency response graphs, note the distinctive bass lift around 50-70 Hz. This spills over into the frequencies of the lowest guitar notes, which is why the amp's low end sounds full. The Bass control lifts the bass response higher.

Mid



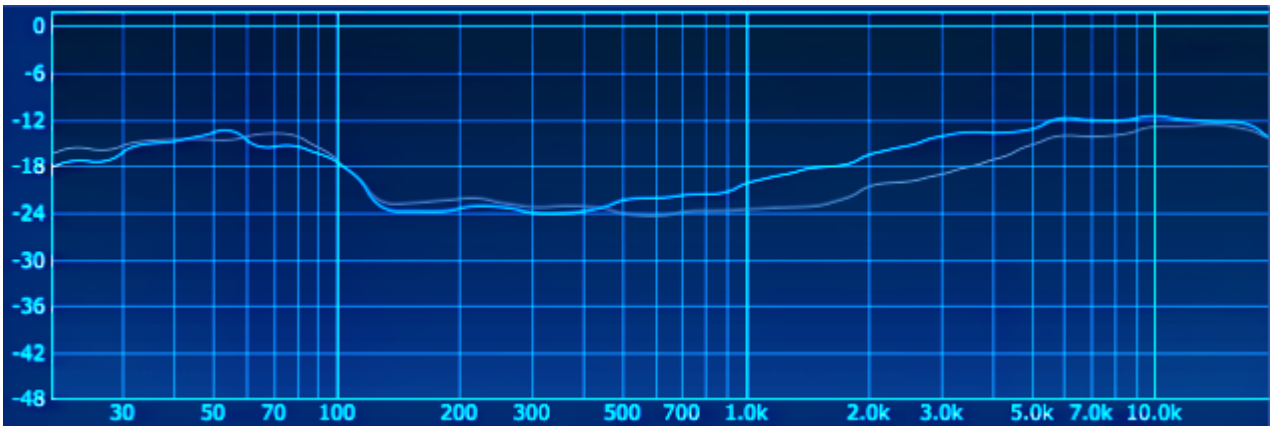
This control is fantastic for shaping tone. At 0.0, Mid gives a strong scoop at 1 kHz. Turn it up for more upper midrange definition and increased high frequency response.

Treble



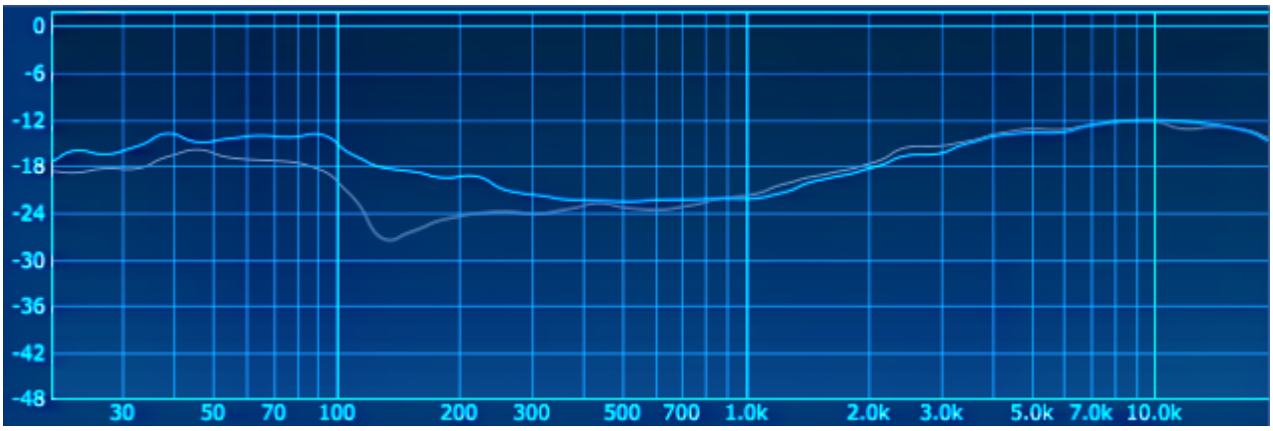
Even at 0.0, you can see the brightness that gives the Elmsley its distinctive frequency response. The Treble control is a serious treble boost—be careful not to turn this up too far.

Presence



The upper midrange lift is in the region that gives definition and intelligibility. I don't know if the Elmsley was designed for amplified acoustic guitar, but the controls work well for that.

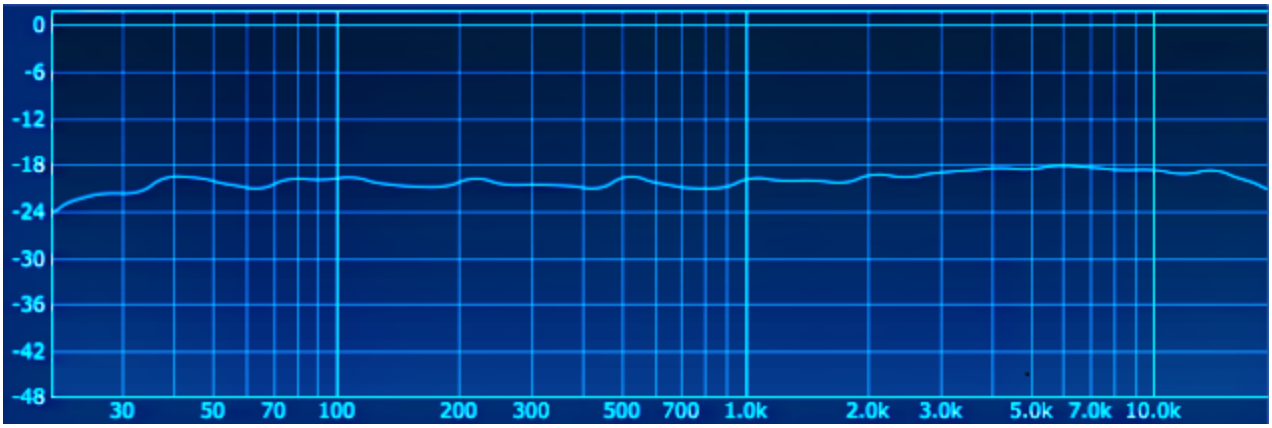
Depth



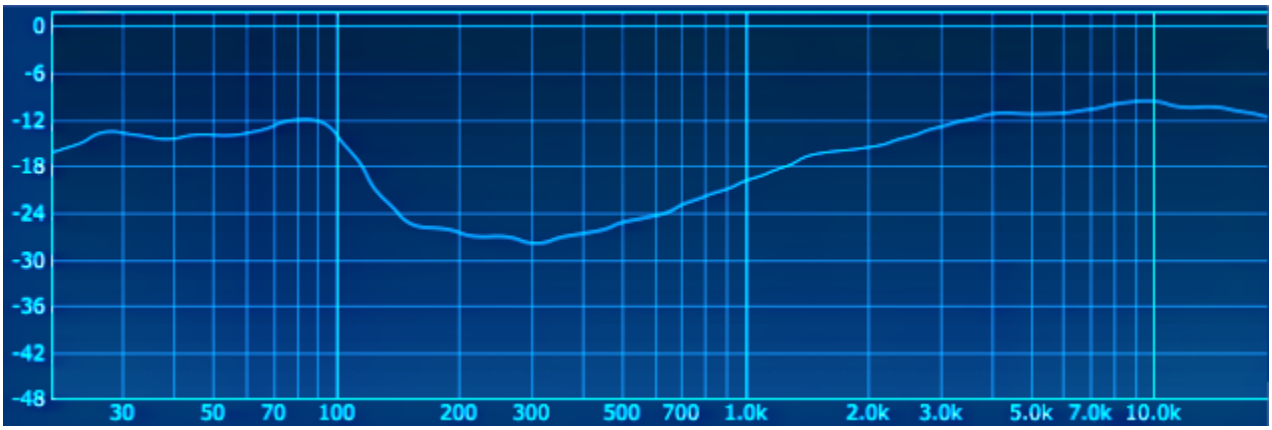
This parameter adds a lift around 100 Hz. It provides an alternative and/or complementary way to shape bass.

NFB (Negative Feedback)

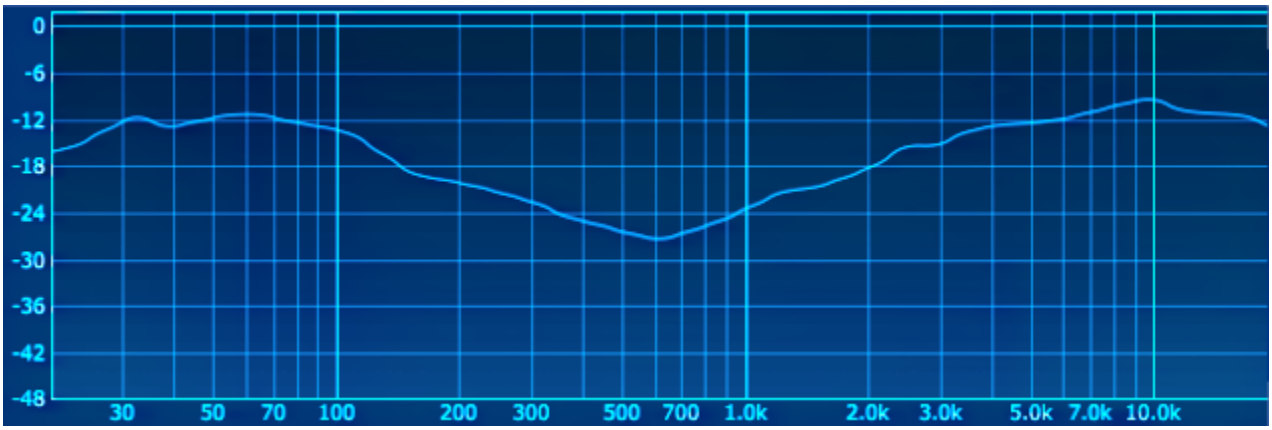
Sound Designer Sam Hwang explains that the NFB parameter alters the feedback within the power section, so you can dial in responses that span from wild and unhinged to tight and punchy. As with many amp designs, the Presence and Depth controls are part of this negative feedback circuit and control the power amp's brilliance and resonance, respectively. Reducing NFB reduces the Presence and Depth controls' influences. When NFB = 0.0, the Presence and Depth controls are effectively deactivated. These interactions provide countless tonal variations and options.



NFB is at 0.0 and all the tone controls are at their defaults. This gives an essentially flat response—it's a great way to hear the guitar's character before making further edits.



This response is with NFB at 1.0 (maximum) at Presence at 10. The result is a strong low end, midrange scoop that's particularly noticeable at the "mud" frequencies around 300 Hz, and an elevated high frequency response.



Here's the response with NFB at 1.0 (maximum) and Depth at 10. Compared to the response with NFB at 1.0 and Presence at 10, the scoop is narrower and shallower, with a higher resonant frequency. The bass response slopes down gently to the scoop, which then slopes up gently to increased treble.

Other Controls

Sag, **Hum**, **Ripple**, **Bias**, **Bias X**, and **Ch Vol** are covered in the section on Common Amp Parameters.

Agua Sledge Bass Amp

Based on the Aguilar Tone Hammer preamp/DI box for bass, the Sledge has bass, treble, and mid (sweepable) tone controls. The main feature is the combination of Drive and Gain controls. Drive does what you'd expect, while Gain kicks in an extra gain structure that ranges from tube-like saturation to hardcore distortion.

Drive

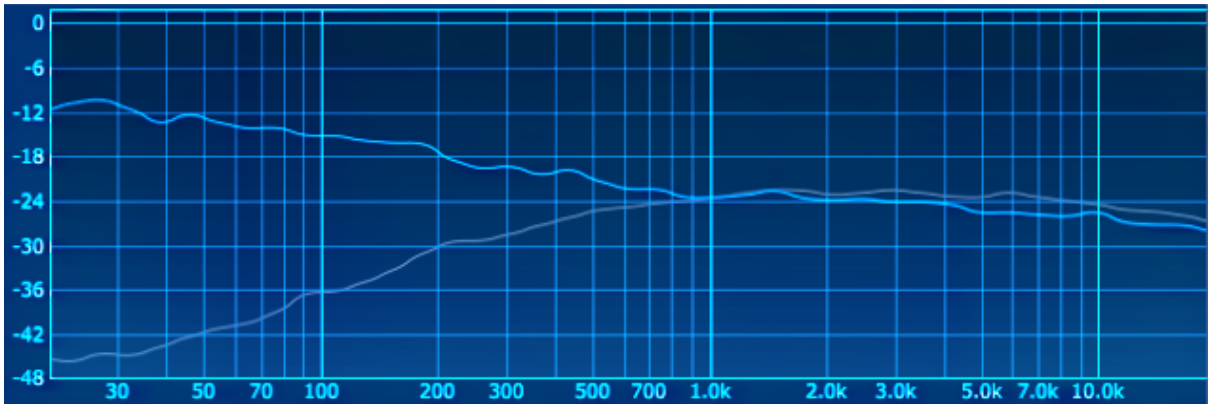
Use this to set an initial saturation level. Even at maximum the saturation is gentle. Drive also establishes the baseline sound upon which you can add Gain to increase the amount of saturation, or go into heavy distortion.

Gain

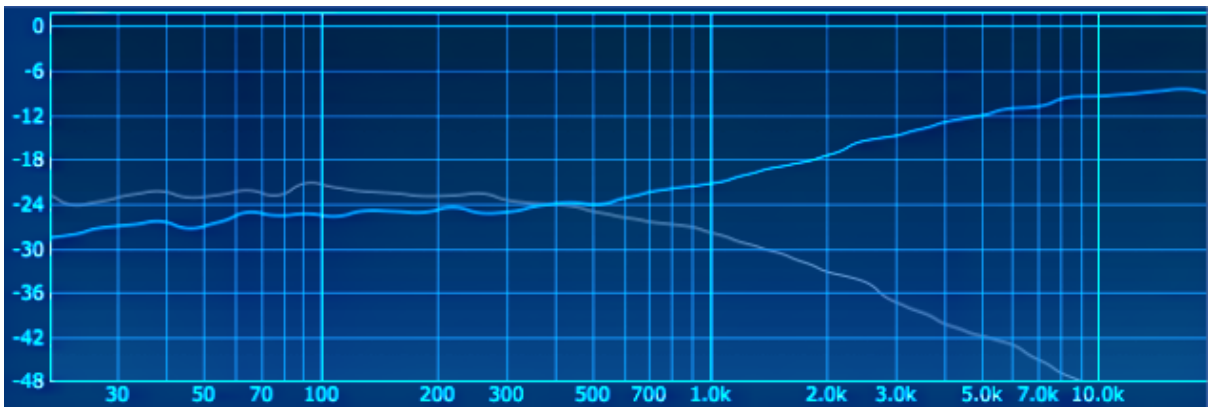
This parameter covers a wider range of potential distortion than Drive. They work together, in the sense that one can fine-tune the other. For example, if Gain at one setting and Drive at 0.0 give the sound you like, you can turn up Drive to increase the distortion. Conversely, Gain can modify the existing Drive setting. The interaction between the two controls, and the ability to assign them to footswitches, offers multiple bass tone variations.

Bass, Mid, Treble Tone Controls

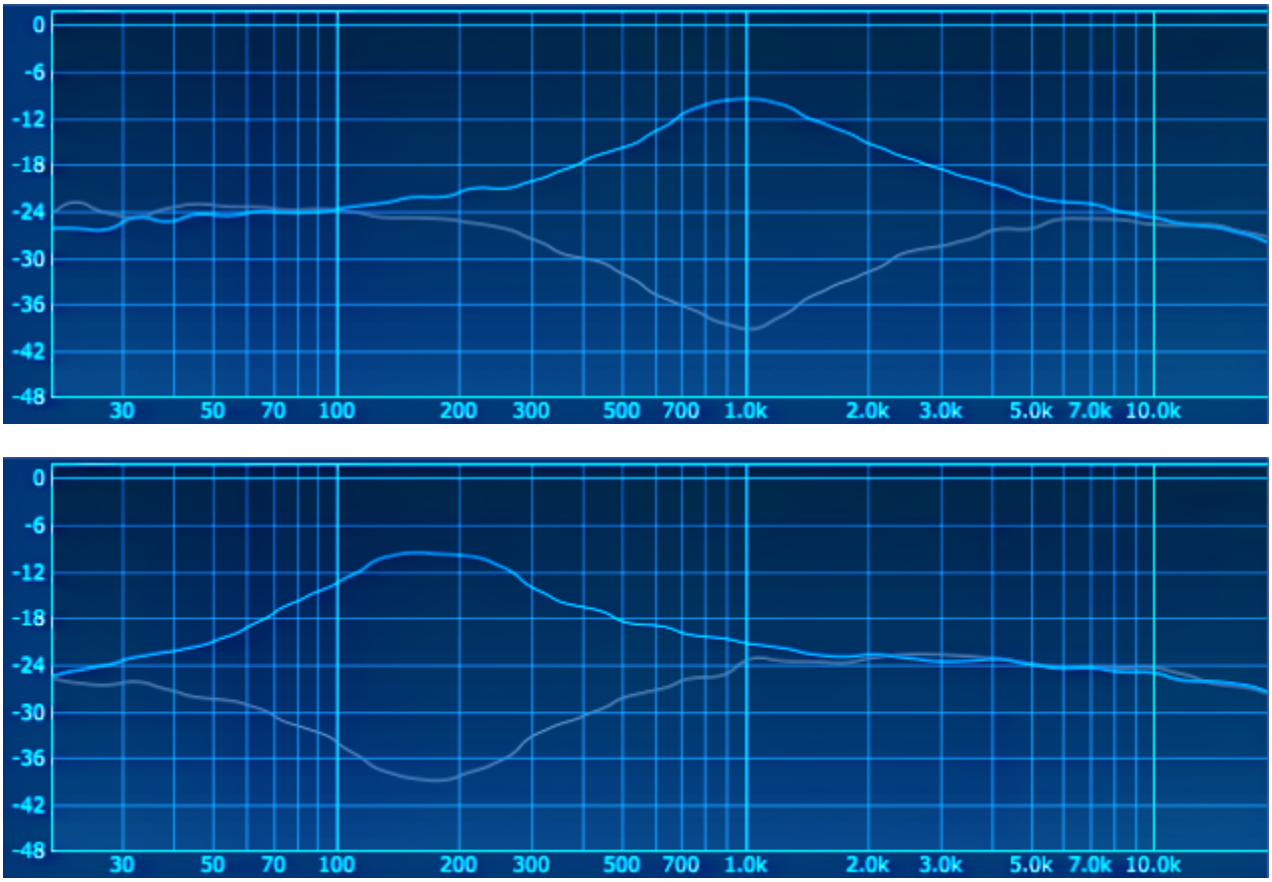
In the tone control frequency response curves, the brighter curve is the control setting at 10.0. The lighter curve is the 0.0 setting. Tone knobs other than the one being measured are in their default positions.



The Bass parameter looks like a conventional shelving control, with a 1 kHz corner frequency.



Treble is the mirror image of the Bass parameter, with a 500 Hz corner frequency.



The upper image shows the Mid control with full boost and full cut. The Mid Freq is at its maximum frequency (around 1 kHz). The lower image shows full boost and cut at the Mid Freq's minimum frequency, which is around 160 Hz.

Master Volume

This also interacts with the Gain and Drive parameters, giving even more options. Also, I've tried the Sledge with rhythm guitar and been pleased with the results, if I followed it with a steep high cut.

Ch Vol

This controls the overall output level. It doesn't influence tone or distortion.

Amp Blocks in Version 3.70, with Presets

V3.70 introduced five Line 6 guitar amps, two guitar amp emulations, and two bass amps. The *Free Files folder > Chapter 2 Presets > 3.70 Amp Presets* folder includes custom presets for each amp. The Chapter 2 Presets folder also has a setlist, **eBook v1.4 New Presets.hls**, with all the new presets that debuted in v1.4 of this eBook (v3.70 amps and Feedbacker).

Each original amp has a Drive and Boost parameter. These work together as a team. The **Drive**'s character increases distortion smoothly from 0.0 to 10.0. Turning down Drive gives excellent clean sounds.

Applying **Boost** adds an instant overlay of distortion. Boost is not a "one size fits all" control but is optimized for each amp. Also, how Boost interacts with Drive varies for each amp. It's well worth taking the time to experiment with the Boost control in each Amp to hear how it affects the sound.

Assigning Boost to a footswitch makes sense for changing from rhythm to lead, or from crunch to heavy distortion. Line 6's Release Notes for v3.70 provide instructions on how to assign the Boost parameter to a stomp switch:

1. Select one of the six Original Amp Design models.
2. Press and hold the Boost knob. Helix/HX jumps to the Controller Assign screen.
3. [HX only: Press PAGE> to show the Learn knob.] Press Learn.
4. Press the desired stomp switch. Helix/HX automatically assigns it. If you like, adjust **Min Value** and **Max Value** to dial in the variation between low and high boost settings.
5. Press HOME to exit. Wonder to yourself "Wait, it's that easy to assign any parameter in Helix/HX to a stomp switch? Why haven't I done this before?"—or—ponder "I'm a power user who knows all the shortcuts. I should assign parameters to switches more often."

The tone controls are more versatile than conventional amps:

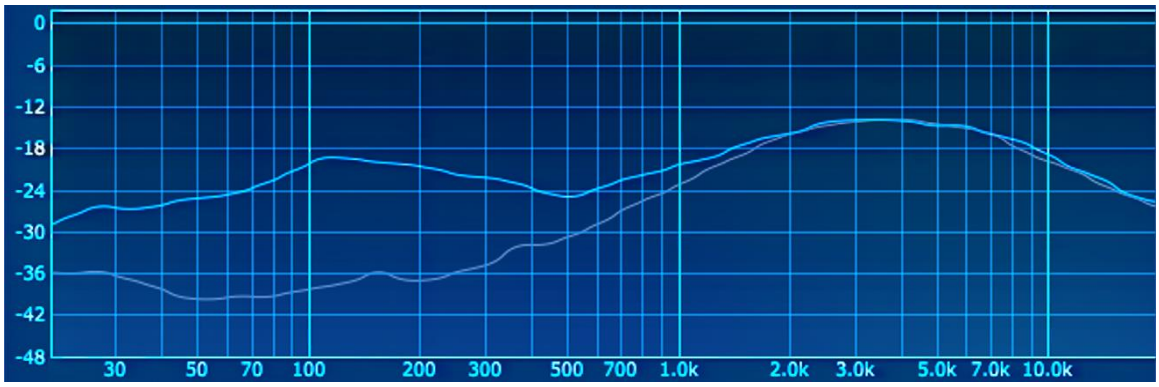
- Many of them "tilt." For example, increasing treble may reduce bass to emphasize the treble increase.
- A control's range covers a wide range of tones.
- The interaction among the tone controls is carefully thought out. You may no longer need an EQ block after an amp because the tone controls can provide the sound you want.
- Despite having the same names, the tone controls are customized for each amp.

The tone control response graphs for each Amp show how a control's response changes over its range. These graphs do not include the Amp's Cab. They relate solely to the Amp. The curves set all controls to 5.0 except for the control being analyzed. Each control shows two curves. The brighter line is the maximum setting (10.0), while the darker line is the minimum setting (0.0).

Line 6 Clarity

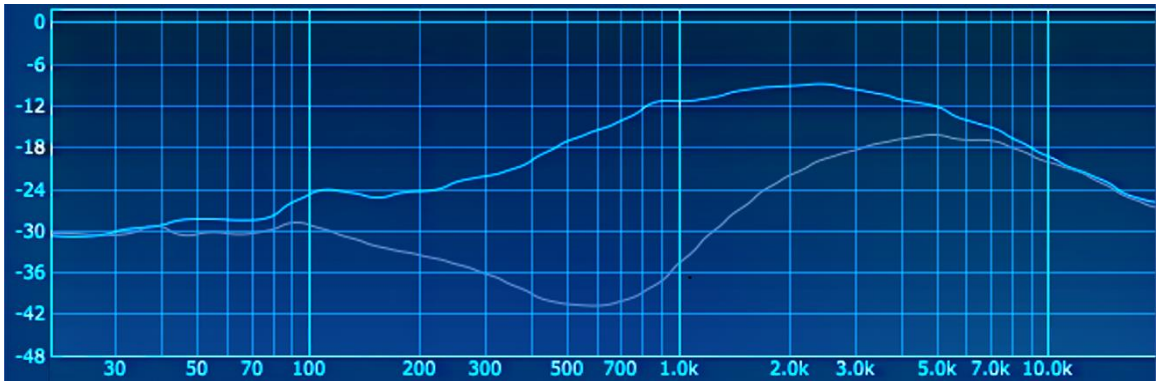
This amp combines a precise solid-state sound with a veneer of tube grit. **Ch1 Drive** is clean for most of its range. Higher Ch1 Drive settings let the tone controls tailor the distortion. For example, you might want an overall clean sound. However, you can add grit in the bass or midrange by boosting those controls.

Bass



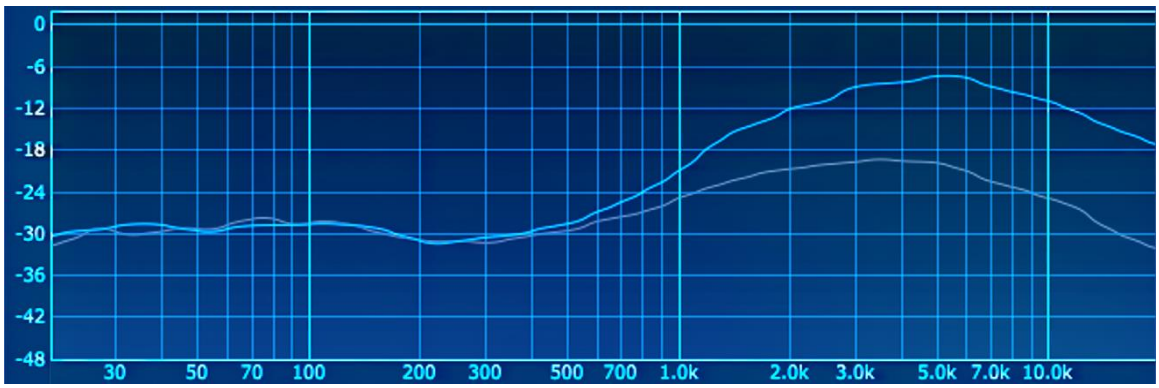
Turning up Bass adds a hump around 100 Hz. This simultaneously creates a shallow scoop centered around 500 Hz. Cutting the bass starts rolling off lows in the midrange. The response gradually levels out below 100 Hz. The extended bass response makes Clarity a potential bass amp candidate.

Mid



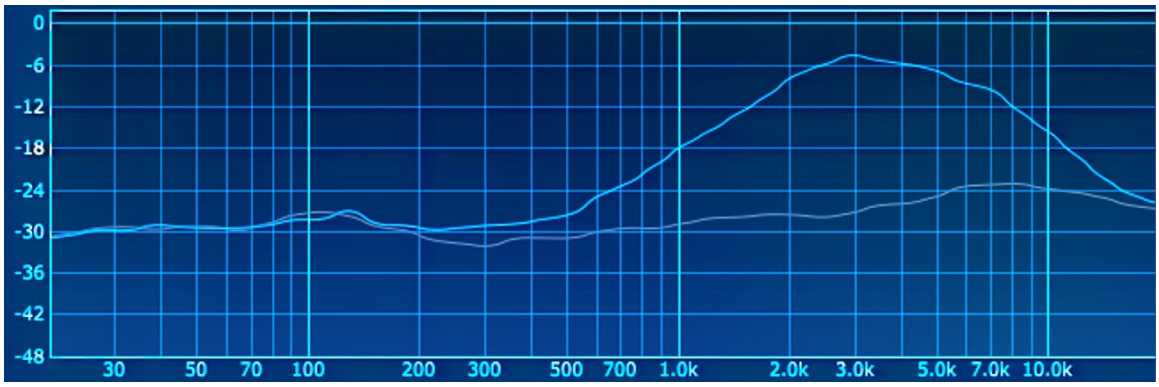
The Mid control does a broad, big boost in both the midrange and upper midrange. Cutting produces the most drastic scoop of any of the new amps. In between these two extremes, you can complement the other tone controls by tailoring the midrange.

Presence



Presence boosts and cuts a decent amount of brightness. It’s ideal for fine-tuning the Mid and Treble control settings. Turn it up for a major treble boost, or turn it down for a little high-frequency shine.

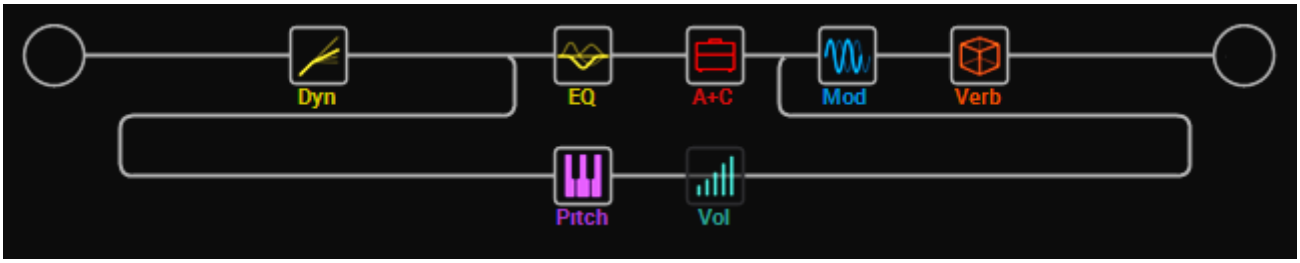
Treble



Clarity has a natural, broad high-frequency lift. Boosting Treble turns the lift into a massive peak that's centered around 3 kHz. Turning Treble down all the way cancels out the inherent high-frequency lift. So, with the other tone controls set to their midpoints, the response is close to flat. This amp works well for giving humbuckers more gloss and brightness than they traditionally deliver.

Clarity Presets

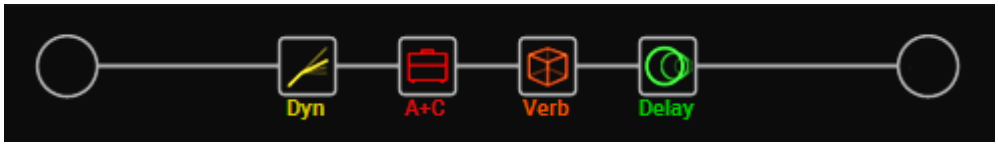
Sweet Jazz Lead



This sweet, mellow jazz lead includes an optional octave-below sound *à la* Wes Mongomery. It's designed for the neck pickup, but that's not essential.

Dynamics: LA Studio Comp applies mild compression, with Emphasis turned down to apply compression equally to high and low frequencies. Aim for -6 dB to -9 dB of gain reduction with single-note leads.
EQ: Enabling this Tilt EQ block tilts the response away from treble, and toward bass, for a round jazz tone.
Amp+Cab: This variation on the Clarity default pulls back on Drive and Bass, but increases Midrange and Presence. Sag is 10.0 to increase sustain. The 2x12 Jazz Rivet Cab swaps out the default close-miked 57 Dynamic mic for an 84 Ribbon mic placed almost 11 inches back.
Mod: The Legacy Analog Chorus adds slow, animated modulation.
Verb: For an intimate jazz club feel, the Legacy Room Reverb is a good choice.
Pitch: This block creates an octave-divided guitar sound by setting the Poly Capo to -12 semitones. Tracking is X-Stable. Choosing 0.0 for Auto EQ preserves the bassy, jazzy sound with higher notes. Although this preset is designed for single-note lines, playing occasional block chords with the Poly Capo enabled adds a thick depth to the sound.
Vol: This Gain block is set to -120 dB. *Bypassing* it enables the octave-lower effect. Enabling it mutes the octave-lower effect. Simply bypassing the Pitch block instead would still let the dry sound through, which isn't ideal.

Afropop



Afropop emulates the bright, flowing guitar tones in highlife and other African genres. Typically, the guitars use single-coil pickups. This preset gives the same kind of brightness to humbuckers. I prefer neck and middle pickup positions. Those using single-coil pickups may want to dial back the Presence.

Dynamics: Trim the input level or threshold to produce about -6 dB to -9 dB of gain reduction. Turning Knee up full gives a gentler compression action.
Amp+Cab: Clarity is perfect, because of its basically clean sound that has a bit of an edge. Compared to the default, there's a little more Ch1 Drive, Treble, and Sag, a lot more Presence, and less Bass.
Verb: Dynamic Ambience gives the feeling of playing in a room, small club, rehearsal hall, etc.
Delay: Bucket Brigade gives that "older pedal" sound and creates a good stereo image that collapses well to mono. The Mix setting is conservative. You may prefer more delay in the mix.

Line 6 Aristocrat

The release notes say this amp's "features aren't limited by what can be done with analog circuits." The tone controls reflect this. As a result, they have more power than you might expect. Start with conservative boosts and cuts. Small changes may be all you need.

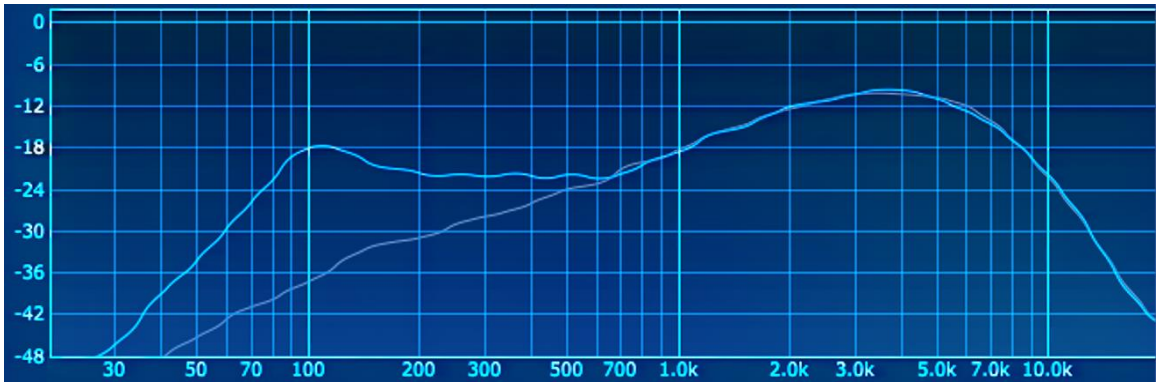
Ch1 Drive and Boost

The Drive control goes smoothly from clean to rich distortion. Settings below 5.0 give a sweet distortion. Increasing Boost retains the sweet character, but increases touch sensitivity and emphasizes the distortion.

This sophisticated amp has a naturally bright character when paired with its default 1x12 US Deluxe cab. For a warmer sound, one option is swapping out the 57 Dynamic mic for a 160 Ribbon mic.

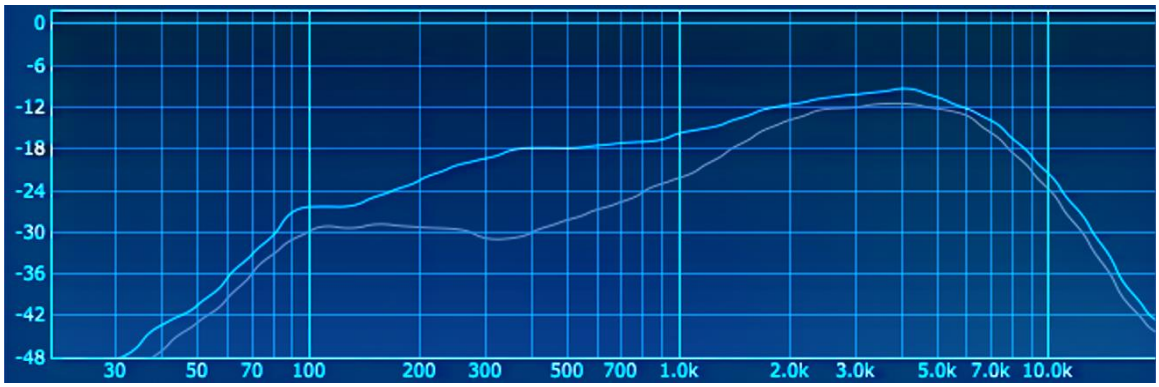
It's best to use a light touch on the controls, because there's much built-in versatility. Even slight changes can make significant tonal differences.

Bass



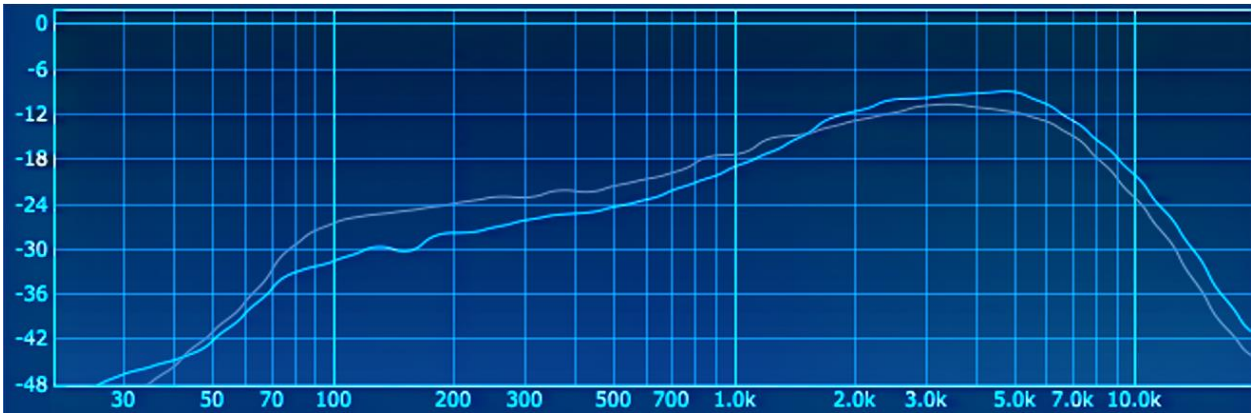
Boosting provides a major hump around 100 Hz, which is around the guitar's lowest range. Cutting "tilts" the bass response downward, which emphasizes the amp's natural articulation peak around 2 to 5 kHz. If you want strong upper mids, consider cutting the bass instead of boosting the upper mids.

Mid



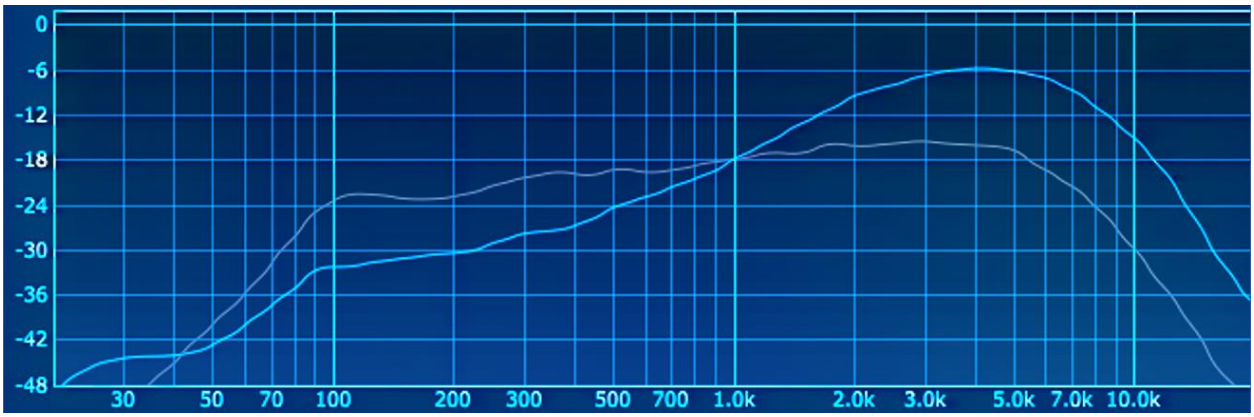
Turning up Mid increases the "warmth" range centered around 300-400 Hz. Reducing the mids tightens the sound and reduces any potential for mud. This also opens space for other instruments.

Presence



At 10.0, Presence gives a slight high-end boost. However, it also reduces the lower mids and upper bass to brighten the overall sound. At 0.0, the high-end lift goes down only a little bit, but the levels of the lower mids and upper bass increase to restore balance.

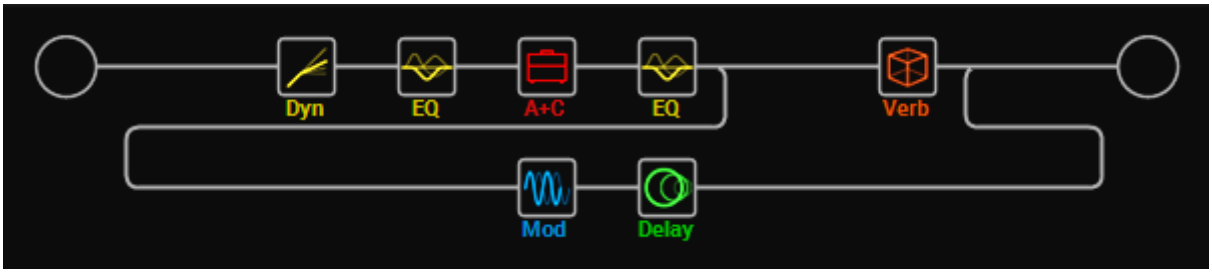
Treble



This tone control tilts the response. Turning up the treble gives a major boost around 3 – 5 kHz, but also tilts the bass response downward starting at 1 kHz. Try this with humbuckers for a tighter, more single-coil sound. Turning Treble down produces a flatter response, with the response falling off around 5 kHz.

Aristocrat Presets

Modern Country



Although designed for modern country and bridge pickups, this also works well for rock.

Dynamics: Adjust the input level or Deluxe Comp Threshold for around -3 dB to -6 dB of gain reduction.
Pre-Amp EQ: To smooth the sound, the Parametric EQ's Mid Gain reduces the response at 4.4 kHz.
Amp+Cab: Compared to the default, there's more Brightness, Presence, and Sag, but less Ch1 Drive. The Cab opts for a 2x12 Double C12N instead of the default 1x12 US Deluxe. Compared to the default 57 Dynamic and close-miking at 1.00", the 421 Dynamic mic and 4.25" distance open up the sound.
Post-Amp EQ: A 1.6 dB Mid Gain boost with a broad Q at 1 kHz brings the sound more forward. The High Gain stage adds a +3.0 dB high-frequency lift starting at 3.1 kHz.
Verb: Nothing special here—just some nice, twangy springs.
Mod and **Delay:** The two function as a team to reproduce the classic Sun Studio slapback echo. The Retro Reel Mod provides the tape sound. The 135 ms mono Simple Delay follows the Retro Reel, because with tape the echo came off the playback head, after it had been recorded. The echo typically fed into a mixer, so the parallel path re-enters the main signal chain after the amp's reverb.

4B.BigAristocrat

When I need a big sound with more definition than the **4B.Gigantic 2203** preset, this gets the call. The two presets are similar. The prefix follows the same naming convention as other multiband presets: **4B** = four-band multiband preset, **dot** after the number of bands = requires a 4-path Helix.

Dynamics: These blocks split the input into High, Mid, and Low frequency bands, using the 3-Band Compressor blocks from the Favorites folder (**3Bd_HiSplitComp.fav**, **3Bd_MidSplitCom.fav**, and **3Bd_LowSplitCom.fav**, respectively).
Pre-Amp EQ: The Parametric EQ cuts at 3.1 kHz to sweeten the high-frequency tone. Bypass it for more bite.
Amp+Cab: Each amp is tweaked for its specific frequency range. The Low and Mid bands use the default mikings and cabs. The High band changes the mic to a 47 Cond FET and alters other settings to for a creamier high-frequency sound.
Vol: These three "utility" blocks simplify editing. Enabling a block mutes its respective band. Muting two bands allows tweaking the remaining band in isolation. After editing the preset as desired, you can delete these.
Post-Amp EQ: The pre-Amp EQ creates a sweeter sound by reducing distortion at higher frequencies, but this also reduces the highs. So, the post-Amp EQ boosts the sweeter sound to compensate for not having as many highs going into the amp. The EQ also increases the output level, and trims Low Gain by -4.0 dB.
Mixer: The Mid and High bands are panned to 50 left and right, respectively.

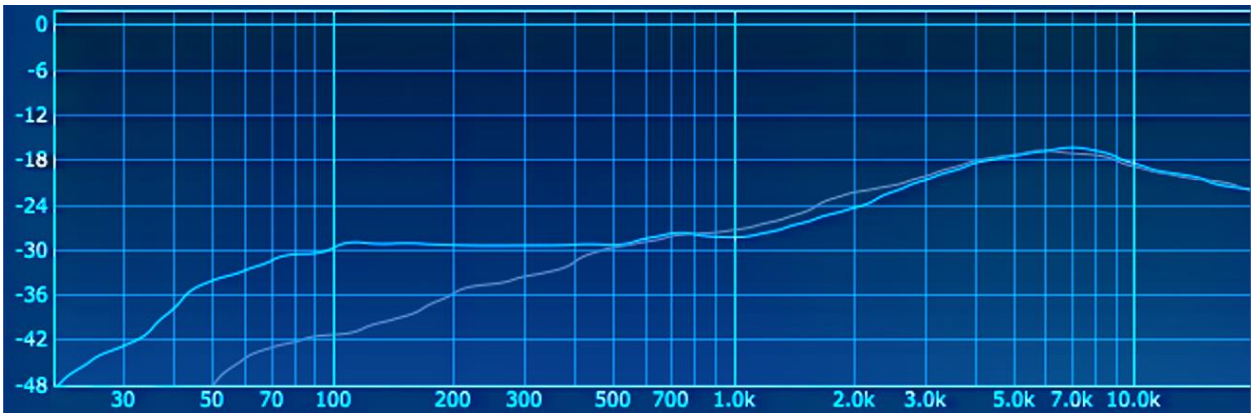
Line 6 Carillon

This strong-sounding amp has a full, midrange-forward character. However, the tone controls are sufficiently flexible that it's easy to emphasize the high frequencies. This isn't my amp of choice for a bassy low end. Its "tight," combo amp feel is naturally bass-shy.

Ch1 Drive and Boost

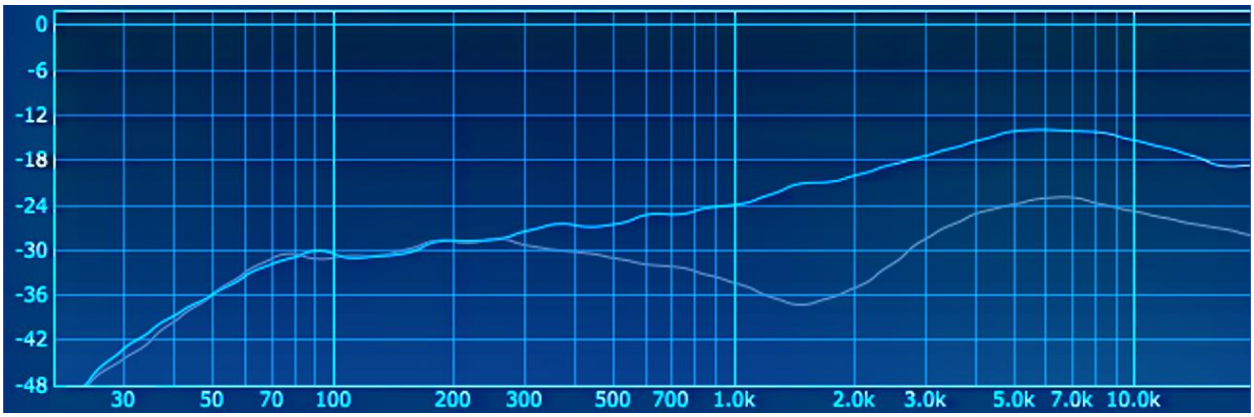
Like the Aristocrat, Carillon cracks the code on the Drive/Boost relationship. Also like the Aristocrat, don't overlook the "clean" sounds that are possible with Drive settings below 5.0. However, unlike the Aristocrat, high Drive settings pour on more distortion.

Bass



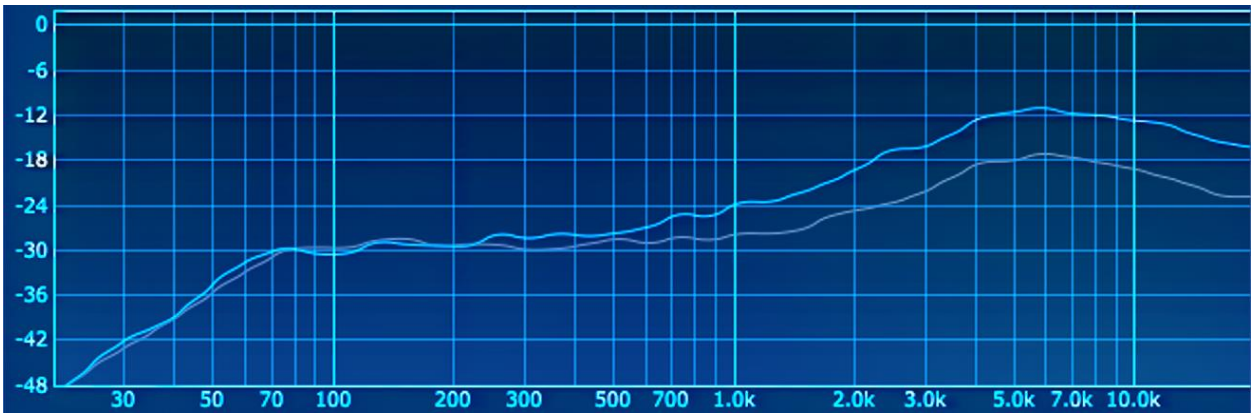
Boosting the bass is more about flattening the response down to 100 Hz, instead of adding a peak. The guitar's low end becomes fuller, but the bass is never overbearing. Cutting rolls off the response in the bass range. This gives a thinner overall tone.

Mid



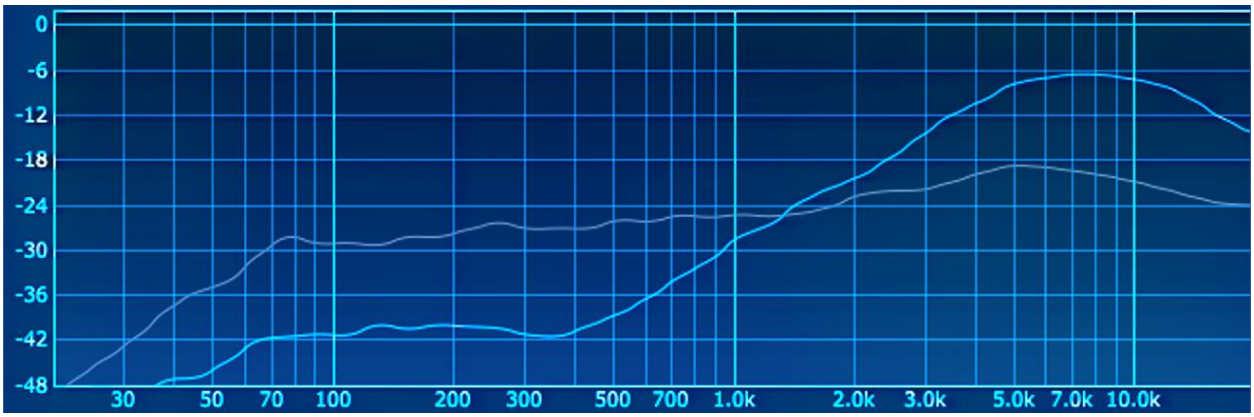
The midrange boost is classic—it covers the range from 500 Hz or so to the upper mids. Cutting produces a deep scoop between 1 and 2 kHz. Normally when boosted this is the range that pushes sounds forward in a mix, so cutting places the guitar more in the background.

Presence



This is a textbook presence control that either lifts or drops the curve that extends from around 500 Hz up to 10 kHz and beyond. It's ideal for brightening up the guitar when you want brightness without harshness.

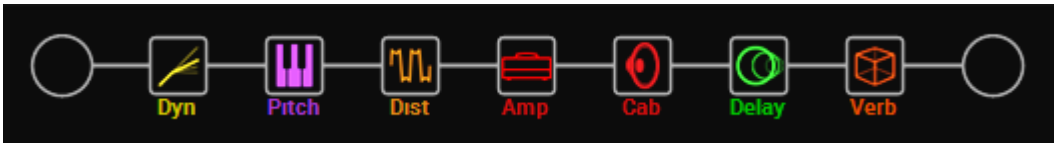
Treble



This is another example of a tone control that tilts response. At maximum, there's a significant high-frequency boost but also a drop in the lower mids and upper bass. Cutting tilts the natural high-frequency boost downward, and retains bass down to 100 Hz. Setting treble to 0.00, and the other controls to 5.00, produces a relatively flat response up to 2 kHz. Above that frequency, there's a mild high-frequency lift.

Carillon Presets

SensitivCarillon

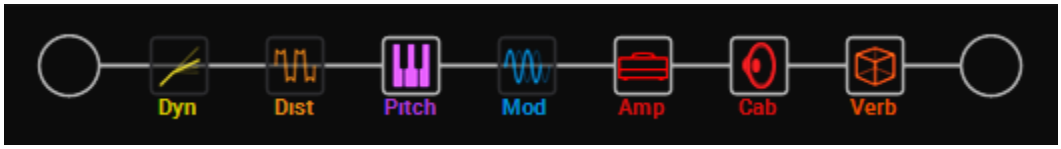


This was made for movie soundtracks…it's 3 AM, and there's a bittersweet moment as the lovers say goodbye for what may be the last time. That's when the slow, evocative lead lines, with lots of dynamics where you alternate between digging into notes or laying back, swells up in the background.

Having Reverse Delay and Dynamic Bloom reverb in series is unconventional. I like using both. You may prefer one or the other, depending on the context.

Dynamics: With the LA Studio Comp, aim for between -6 dB and -9 dB of gain reduction. **Tip:** This preset lends itself well to experimenting with different Dynamics blocks.
Pitch: The Pitch Wham allows the pedal to bend up 2 semitones in the toe position. The heel position has no pitch change. Bending electronically produces a different character than bending strings.
Distortion: To enhance the sound further down the chain, the Regal Bass DI is ideal.
Amp: The Amp is close to the default, other than more Presence, more Sag, less Drive, and optimized Bias and Bias X settings.
Cab: The Dual Cab uses the default 2x12 Blue Bell and 4x12 Greenback 30.
Delay: Reverse Delay complements the Bloom Reverb by creating ambient, enveloping sounds that occur in the spaces between notes.
Verb: I love the Dynamic Bloom reverb. It's why you need to play slow, languid leads. Without spaces between notes, there's no way for the reverb to bloom.

Brit Surf Garage



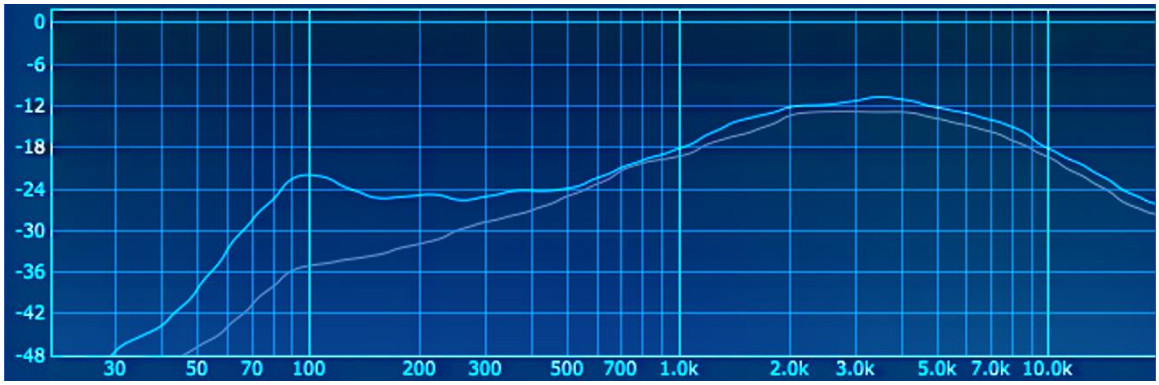
This preset loads with Dynamics, Distortion, and Tremolo bypassed. The combination of effects you choose leans the sound more into a garage, surf, or English invasion vibe. The preset is optimized for the treble pickup. (Extra credit if you use a Mosrite or Burns guitar 😊.)

Dynamics: To increase sustain, enable the Deluxe Comp but note that this preset requires a suitable input level. Aim for -6 dB to -12 dB of gain reduction. If you lower the Threshold, increase the Level. The average level should be about the same whether you enable or bypass dynamics.
Distortion: The Tone Sovereign gives a modern sound to this preset. To go further back in time, substitute the Legacy Buzz Saw distortion. This emulates the Maestro Fuzz-Tone that was favored in much 60s music.
Pitch: The Pitch Wham is your virtual vibrato tailpiece. Push down on the pedal to drop 2 semitones in the toe position. Because this preset is intended mainly for solos, there's no need to burn up DSP with the Poly Wham.
Mod: It's against the law to create this kind of preset without having a tremolo available.
Amp: Compared to the default, the Carillon increases Drive, Treble, Presence, and Sag, but pulls back on Mid.
Cab: The Dual Cab combines the Carillon's default 2x12 Blue Bell (but increases the mic distance to 11.25") with a 1x12 Open Cream that's modified considerably from the default. Like other presets with a Dual Cab, one cab pans left and the other cab pans right to create a stereo image.
Verb: It's a spring reverb. Of course.

Line 6 Voltage

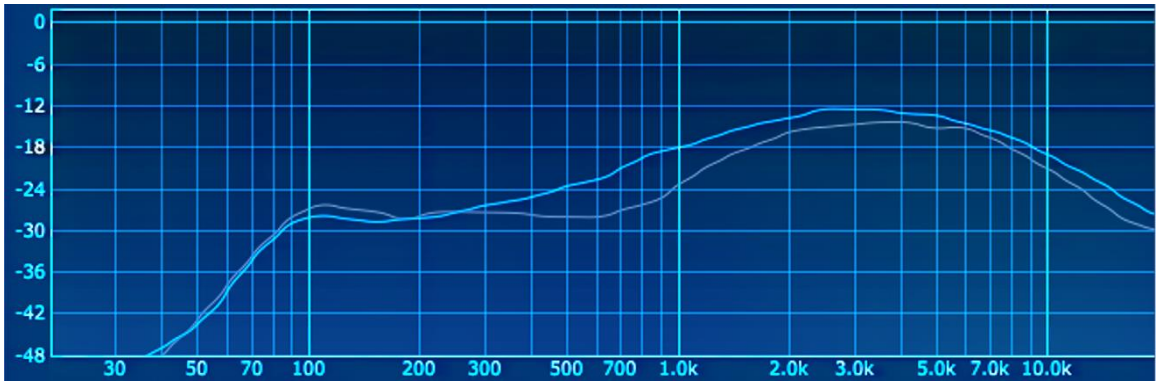
Line 6 calls this “inspired by the classic British ‘plexi’ amp.” Its character is brash, assertive, and rock-friendly.

Bass



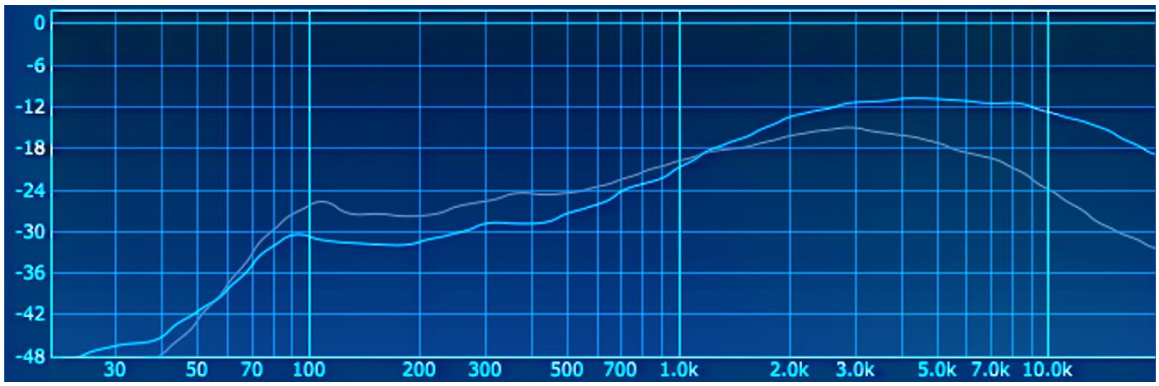
The bass boost gives a strong peak at the guitar’s lowest range, which balances out the inherent high-frequency emphasis. Even with bass at minimum, there’s more bass than the Carillon. It’s on a par with the Aristocrat.

Mid



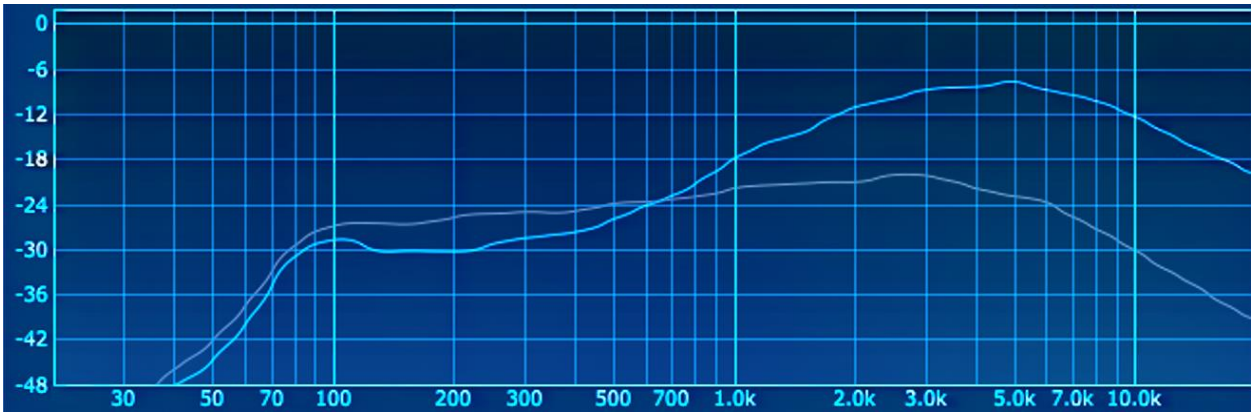
The Mid control is fairly subtle. When boosting, it pushes a little more energy into the 500 Hz to 2 kHz range. Cutting isn’t about a pronounced scoop, but flattens the response between about 100 and 700 Hz. The net result combines a neutral bass response countered by the high-frequency emphasis.

Presence



Again, this control has a tilting quality. Increasing Presence fills out the high end and reduces response below 1 kHz. Yet it retains a slight bump around 100 Hz to keep the low end intact. Cutting presence rolls off the highs for a sweeter sound, and makes sure the boost around 100 Hz remains prominent.

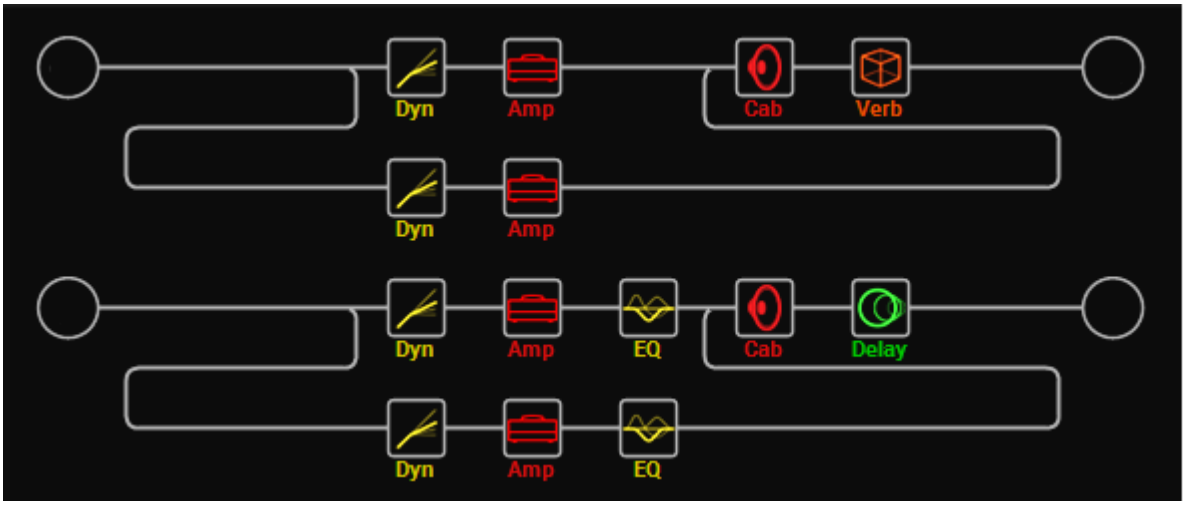
Treble



The Treble control is highly effective. Sure, it boosts highs, but it even maintains the low-end peak around 100 Hz. When cutting, there's a gentle rolloff starting around 3 kHz. This can reduce any potential harshness, and make the sound more easily adaptable to different cab/mic combinations.

Voltage Presets

4B.VoltagPwrChrd



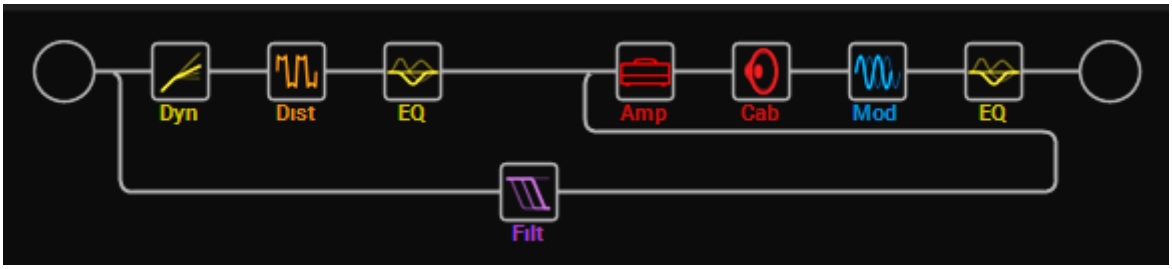
This isn't just a Voltage preset, it's a high-voltage preset. The prefix has the same naming convention as other multiband presets: **4B** = four-band multiband preset, **dot** after the number of bands = requires a 4-path Helix.

Like my other multiband presets, this preset is intended more as a drop-in replacement for amps than an effects chain. However, there is enough DSP for reverb and delay. If you delete those, you can add blocks like Compressors or Distortion at the two inputs, or additional EQs to shape the sound.

This preset collapses well to mono for the stage, but it reaches its full potential with stereo.

Dynamics: These blocks split the input into Low, Low Mid, High Mid, and High frequency bands. The splits use the 3-Band compressor blocks from the Favorites folder (**4Bd_LoSplitComp.fav**, **4Bd_LoMidSptComp.fav**, **4Bd_HiMidSptComp.fav**, and **4Bd_HiSplitComp.fav**, respectively).
Amps: Each amp has slight tweaks to accommodate the different frequency bands.
Post-Amp EQs: These "tune" the responses to the frequencies the amps cover. The choice of parameter values is highly subjective, so feel free to play around with them.
Cabs: The mixer takes advantage of the Dual Cabs by panning each of the four frequency bands to its own cab and miking. As with the post-Amp EQs, the chosen parameter values are highly subjective. Changing even one cab or miking setup can give a completely different sound.
Verb: The Legacy Hall reverb adds ear candy to the Low and Low Mid bands.
Delay: Similarly, the stereo Simple Delay enhances the stereo image of the Hi Mid and High bands.

VoltageWahLead



This incorporates the Autofilter in a lead preset. Much to my surprise, it works. The result is expressive because the filtering ties into your dynamics. The Autofilter input is at the beginning, so it picks up the full dynamic range of your playing. However, it's in parallel with compression and distortion, which creates a richer sound than the Autofilter by itself.

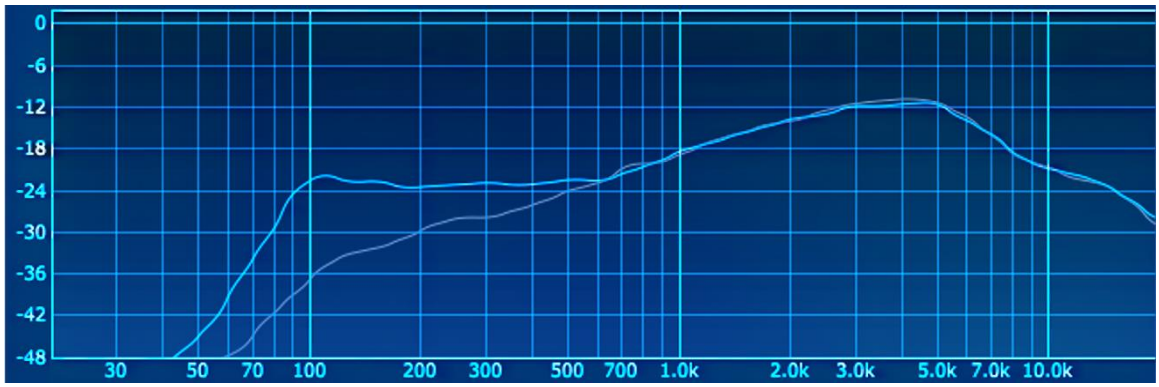
The Autofilter output goes directly into the Amp. This particular setup doesn't have the ear-splitting nastiness that sometimes happens when placing a filter in front of distortion. The Dual Cabs and Ampeg Liquifier chorus deliver a great stereo image, but the preset still collapses well to mono.

Dynamics: The Deluxe Comp helps create a sound with steady dynamics to fill in for the Autofilter when its dynamics vary. You don't need a lot of compression. I aim for around -6 dB to -12 dB.
Distortion: The Obsidian 7000 sounds great in this application. I suppose other distortions would work, but after loading this in I didn't feel the need to go any further.
EQ: The parametric drops a notch around 3.3 kHz to reduce the harmonics generated from this frequency range. The result is a sweeter tone.
Amp: Aside from the Bias, Bias X, Hum, and Sag settings, the parameters are the same as the default.
Dual Cabs: One is a 1x10 US Princess with default settings. The other is a 4x12 XXL V30 whose settings have almost nothing in common with the default.
Mod: The Ampeg Liquifier is a thing of beauty, and it works well with leads.
EQ: The final Simple EQ bumps up the brightness a bit.

Line 6 Kinetic

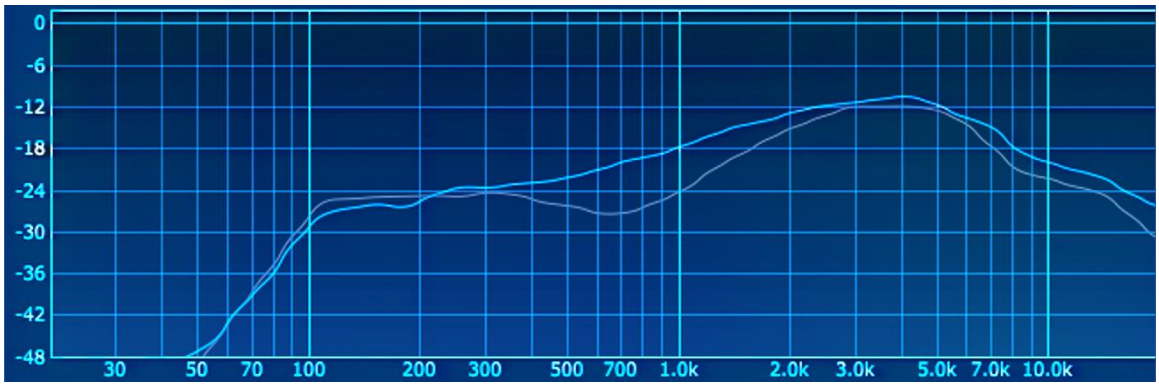
According to Line 6, the preamp stages clip at roughly the same time, so the amp can go from mostly clean to heavily distorted with the guitar's volume control. Turning up Boost adds an emphasis around 700 Hz. This pushes the sound more forward. The amp has the more jagged and aggressive sound of “post-Marshall” rock.

Bass



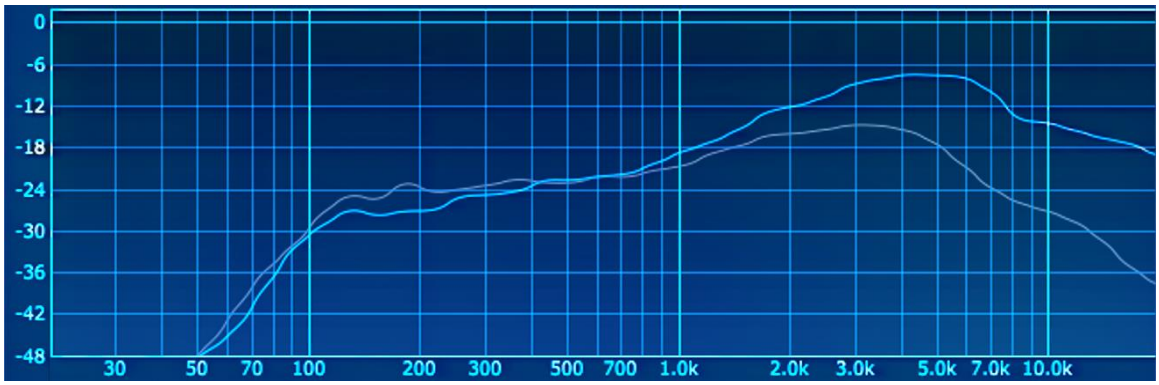
Turning up bass gives the low end needed to “anchor” the more manic distortion effect. Instead of rolling off below 700 Hz, the bass response flattens out. It ends with a slight boost at 100, then a steep rolloff below that.

Mid



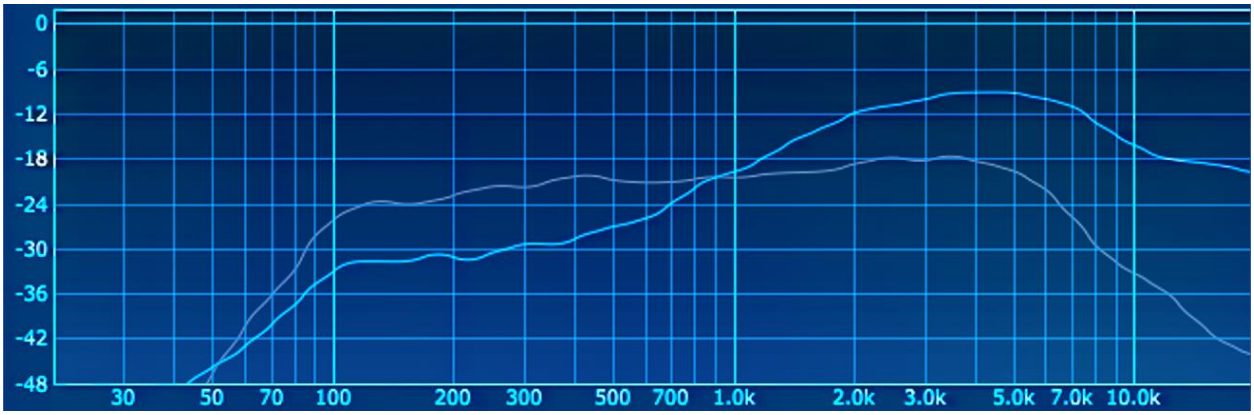
Boosting lifts the response in the 400 Hz to 2 kHz range higher, while cutting creates a mild scoop centered at 700 Hz. When cutting, pushing Boost up full offsets the reduced level at 700 Hz somewhat.

Presence



This amp has a naturally bright high end (the Bass control graph shows what happens above 1 kHz). Presence is like a subtler version of the Treble control, with a focused boost from 2 kHz to 7 kHz. Cutting reduces the amp's naturally bright response with a gentle rolloff that starts around 4 kHz.

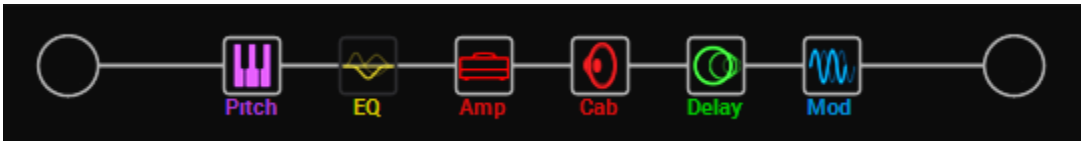
Treble



Turning up Treble gives a broad boost from 1 kHz to 10 kHz. Cutting flattens out the midrange response until reaching a treble rolloff (steeper than the Presence control) starting at 4 kHz.

Kinetic Presets

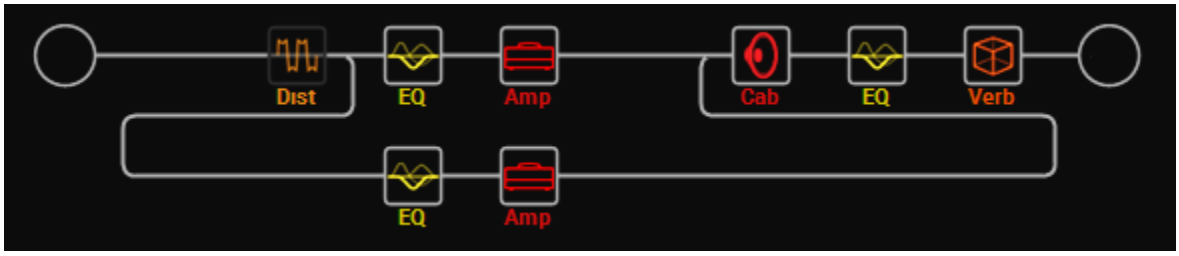
Kinetic 5th



This preset layers the tonic with a fifth below, complemented by stereo delay and flanging. The result is a huge sound that also collapses well to mono.

Pitch: Mixes the original input with the same input dropped down -5 semitones. Auto EQ is 0.0 to avoid brightening the transposed sound.
EQ: The preset defaults to bypassing this block. When enabled, it reduces the Mid and High frequency ranges by -12 dB for a mellower, less bright sound.
Amp: Compared to the default settings, there's less Drive and Mid, more Presence, Treble, and Boost, and a lot more Sag.
Cab: The Dual Cab provides a stereo image. The 4x12 Uber V30 is like the default, except with the mic positioned closer to center. The other speaker is a 2x15 US Dripman that uses the default settings.
Simple Delay: The short stereo delay emphasizes the stereo spread.
Analog Flanger: This Legacy effect is a "because we had enough processing power left over to insert it" effect. I like the flanging, but feel free to bypass it.

Kinetic Rhythlead



As a rhythm preset, the super-wide stereo image is great for LCR (left-center-right) mixes. It also leaves room in the center for bass, kick, snare, and vocals. Enabling distortion gives a strong, bright lead sound.

Distortion: The Pillars distortion works well with this preset. The preset defaults to bypassed distortion. Enable it to go from rhythm to lead.
Pre-Amp Parallel EQs: The pair of 10 Band Graphic EQs pass alternate bands to the two Amps. This enhances the stereo effect and widens the image. They also reduce treble somewhat to tame potential harshness caused by distorting the high frequencies.
Parallel Amps: Compared to the default Kinetic amp, these have more Ch1 Drive because they don't get as much input from the filters. There's also more Presence and Sag.
Cab: The Mixer uses its internal panning to send each amp to its own Cab. Using Dual Cabs further enhances the stereo effect. The two cabs are quite different. One is a 4x12 Mandarin EM with a 160 Ribbon mic, the other is a 2x12 Match H30 with a 57 Dynamic mic. As with most other Dual Cab-based presets, the Cabs are panned left and right.
Post-Amp EQ: The simple EQ adds 10 dB of high-frequency gain. This compensates for the reduced brightness going into the amps due to the pre-Amp parallel EQs.
Verb: Dynamic ambience fills in some room sound.

Line 6 Oblivion

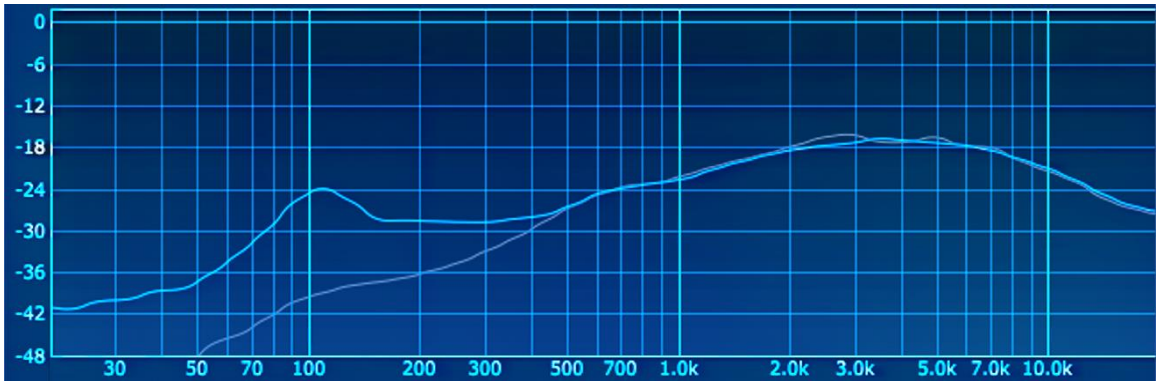
This is version 3.70's gift to metal, thrash, and industrial. Nonetheless, as the preset **Oblivion Smooth** shows, inserting EQ before the amp can give smooth leads and focused power chords. Of the six 3.70 original guitar amps, this one has the most radical tone control possibilities. That's important for shaping the tone of a no-holds-barred kind of amp.

Unlike the other 3.70 amps, it's not possible to obtain a clean sound by turning down the Drive control. It starts pouring on distortion around 1.0. However, settings below 1.0 can give some tasty distortion sounds.

Because of the heavy distortion, Oblivion is well-suited to inserting EQ before the amp and trimming the frequency ranges that are subject to heavy distortion.

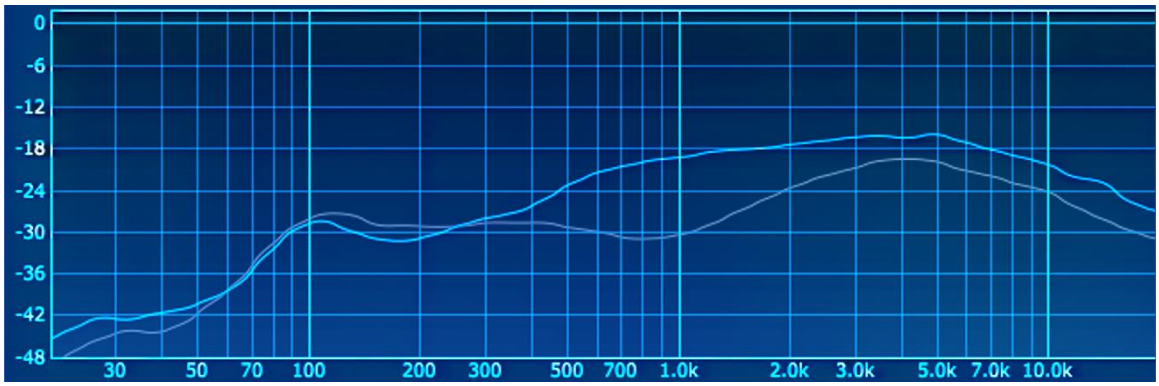
Tip: Set Ch1 Drive to 5.0. Replace the Parametric EQ *before* the Amp with the 10 Band Graphic. Set all band levels to -15 dB. Experiment turning up different combinations of faders. You'll hear a wide variety of sounds.

Bass



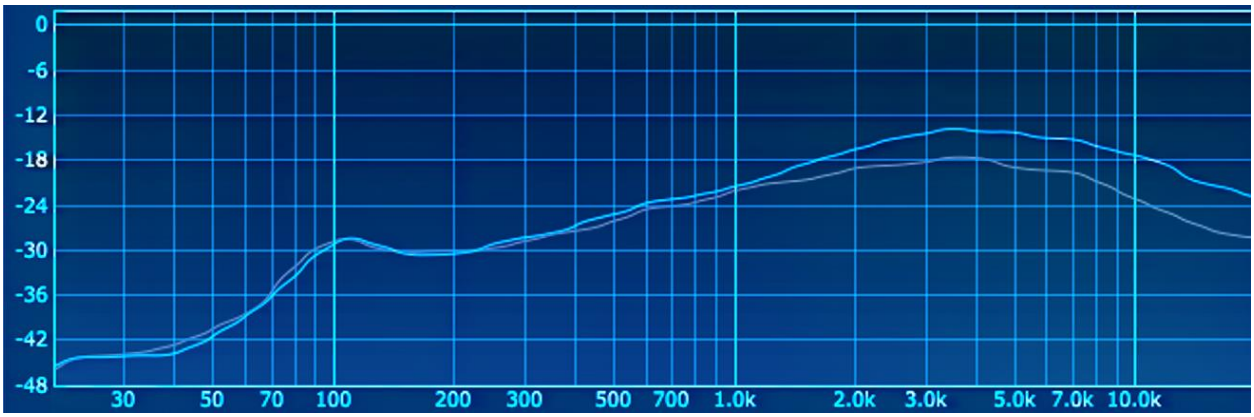
Oblivion's Bass control has the most pronounced bass bump of any of the new amps. Turning this up adds a heavy, weighty quality to the sound. When cut, the response rolls off starting around 500 Hz.

Mid



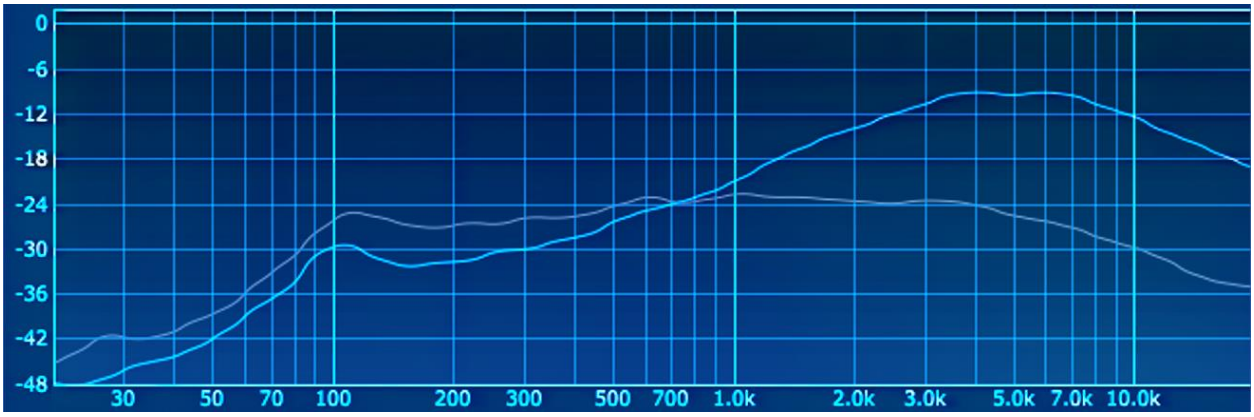
Boosting gives a bulge in the 400 Hz to 6 kHz range. You don't have to turn this control too far up to obtain a meatier midrange sound. Cutting gives a significant scoop around 700 Hz – 1 kHz.

Presence



This is the gentlest of Oblivion's controls. The Treble control allows for the high end to dominate, so boosting Presence is more about adding a brighter quality. Cutting reduces brightness equally subtly.

Treble

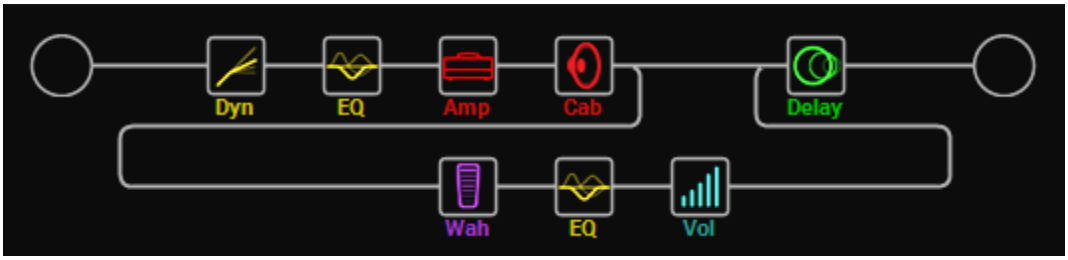


Turning up treble gives a major boost from 1 kHz to 10 kHz, and peaks around 5 kHz. A slight boost adds articulation, but too much creates harshness. Cutting rolls off the naturally bright sound gently, starting around 4 kHz.

The way Treble and Presence work together is a good example of the intelligent interaction among controls. Turning down Treble and turning up Presence gives brightness without harshness. Also try turning down Presence, then turning up Treble for the desired brightness.

Oblivion Presets

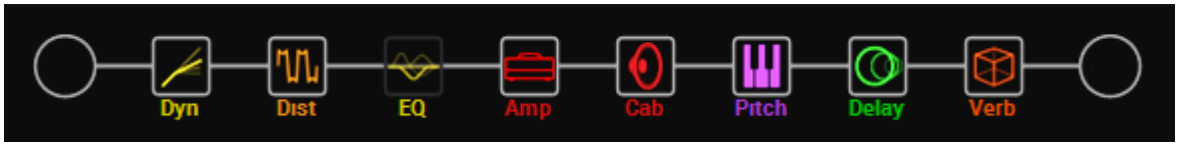
Oblivion Smooth



This preset cleans up Oblivion's aggressive tone to give "singing," single-note leads. The preset loads with the Wah bypassed. Assign the Vol block Bypass to a footswitch so you can bring the wah in or out.

Dynamics: The Red Squeeze increases sustain so that the lead "sings" more.
Pre-Amp EQ: Parametric EQ is key to smoothing the sound. It cuts -12 dB at 1 kHz, using fairly narrow Q. If you bypass this block, the sound is very different.
Amp: This is close to the default, but increases Treble and Presence, while reducing Bass. There's more Sag to extend the sustain, and no hum.
Cab: The Dual Cab uses the default 4x12 Cali V30 and 1x12 Cali IV.
Wah: For some optional wah action, use the paralleled Vetta wah.
Post-Wah EQ: Because the wah takes over at higher frequencies, this EQ places a broad notch at 2.9 kHz to keep the upper midrange under control.
Vol: Enabling Vol (the default state) sets its gain to -120 dB, which prevents the wah sound from reaching the output. This is better than bypassing the wah, which would send more post-Cab sound to the output and increase the level. Vol could also alternate between two amounts of gain to bring in more or less of the wah sound.
Delay: To give a spacey vibe for slower leads, the stereo Simple Delay generates echoes at 250 ms for the left channel, and 200 ms for the right channel. Bypass the Simple Delay for a more direct, focused sound.

Oblivion Bite



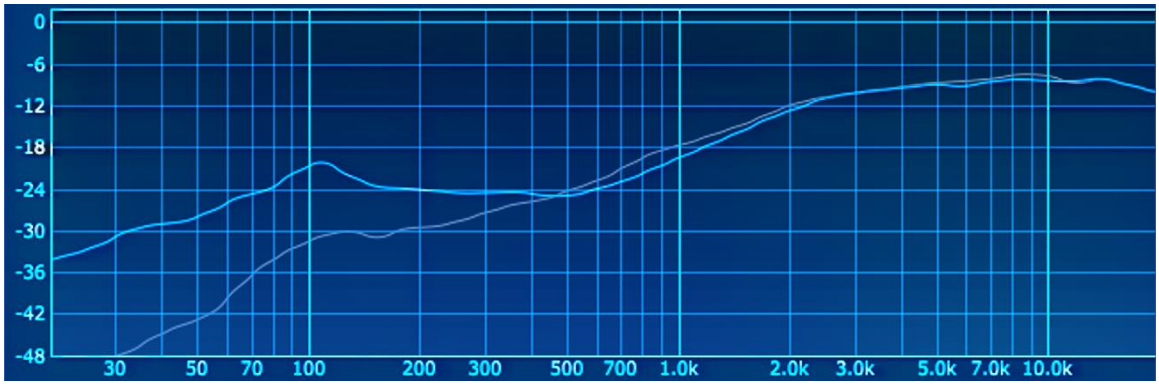
This preset has more bite than the Oblivion Smooth preset. Featuring a wide stereo image that collapses well to mono, this sound is a natural for recording but is also suitable for live performance.

Dynamics: The LA Studio Comp increases sustain. Edit the Peak Reduction control for around -3 dB to -6 dB of gain reduction.
Distortion: Scream 808, an oldie-but-goodie, enhances sustain.
EQ: The preset loads with this Parametric block bypassed. Enable it to add a notch at 1.8 kHz, and a High Cut to reduce response above 4 kHz. This creates a slightly sweeter tone that's particularly noticeable when using the bridge pickup.
Amp: Compared to the default, there's a little less Ch1 Drive, less Bass, no Presence, more Mid, and a bit more Sag to extend sustain.
Cab: This Dual Cab modifies some 4x12 Cali V30 settings and adds a modified 4x12 Greenback 30 for the other Cab.
Pitch: Because this is a solo-oriented preset, there are no nuances with the pitch parameter. It's controlled by EXP 1, and solely about providing a one-octave dive bomb.
Delay: A short delay time with the stereo Simple Delay widens the stereo image. Leave this enabled even when feeding a mono system, because its settings contribute to the tone in mono as well as stereo.
Verb: The Hot Springs gives some vintage vibes.

Brit 2003

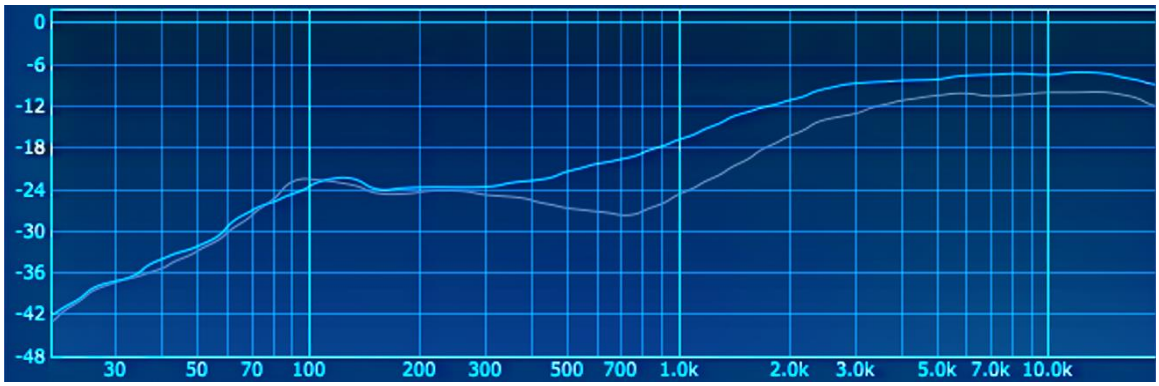
Unlike the new Line 6 amps, the tone controls in conventional amps typically just do “more” or “less” within their specific frequency ranges. So, there’s less interaction among the tone controls.

Bass



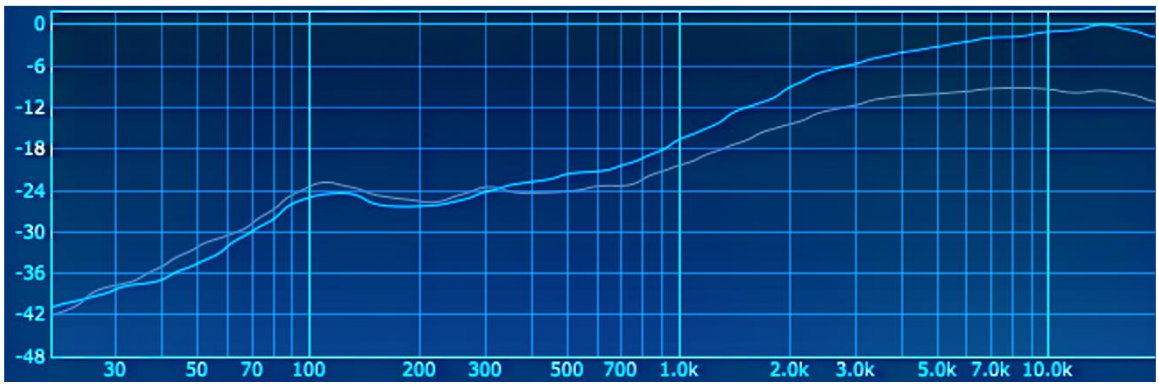
Boosting increases bass starting around 500 Hz, while producing a significant bump around 100 Hz. Cutting rolls off starting around 500 Hz. However, there’s still a little bit of a bump left, even at the lowest setting.

Mid



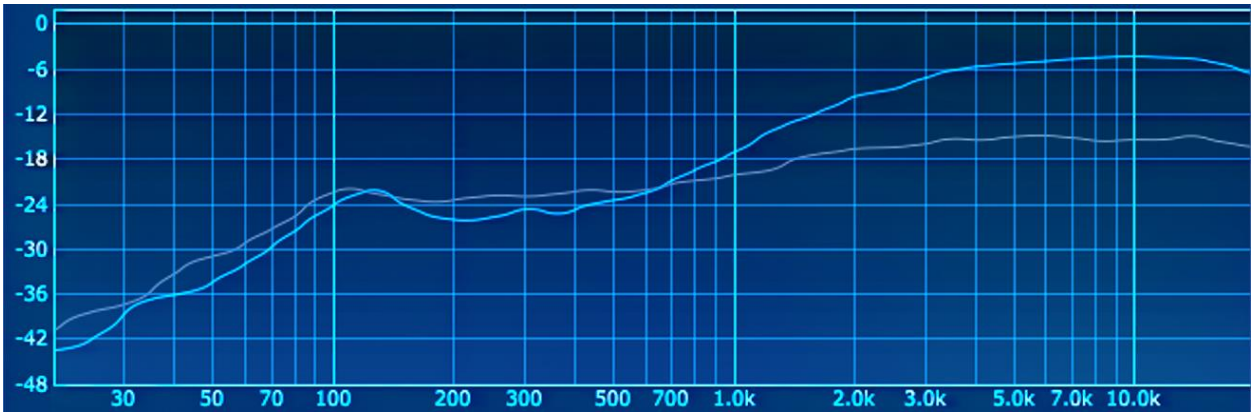
Mid boosts predominantly in the range from 300 Hz to 2 kHz, but also extends into the treble range. Cutting creates a scoop around 700 Hz that extends from 300 Hz to about 5 kHz.

Presence



Turning up Presence lifts the response from 300 Hz to beyond 10 kHz. Turning it down all the way trims the treble in this range.

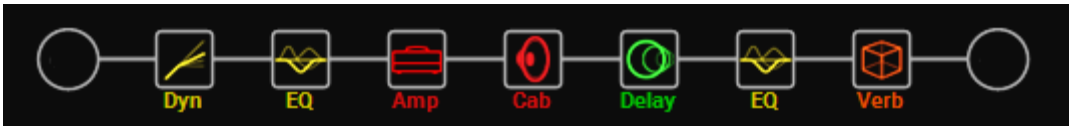
Treble



Like Presence, this is what you'd expect: turning it up gives a major boost in the 500 Hz to 10 kHz range, while turning it down trims the response over the same range.

Brit 2033 Presets

Lead 2203



This is a straight-ahead rock lead preset. There's enough processing power left over for adding more/different effects, including most stereo ones.

Dynamics: The Rochester Comp levelling amplifier provides sustain at the front end.
Pre-Amp EQ: A Simple EQ trims the high frequencies to reduce harshness at the amp.
Amp: The main Amp differences compared to the default are less Bass, slightly more Treble, and more Sag.
Cab: In the 4x12 1960A T75, the 7 Dynamic replaces the 57 Dynamic for a smoother sound and positions it closer to the edge. The Distance is 8.00" instead of 1.00". The 4x12 Greenback 25 Cab uses the default settings.
Delay: A stereo Simple Delay adds a subtle 250 echo and creates a stereo image.
Post-Amp EQ: The pre-Amp EQ creates a sweeter sound by reducing distortion at higher frequencies, but this also reduces the highs. So, the post-Amp EQ boosts the sweeter sound to compensate for not having as many highs going into the amp.
Verb: The Stereo Hot Springs block enhances the stereo image. It also adds the unique spring sound to the lead.

4B.Gigantic 2203

This is becoming my favorite power-chord preset for huge, thick sounds. The prefix follows the same naming convention as other multiband presets: **4B** = four-band multiband preset, **dot** after the number of bands = requires a 4-path Helix.

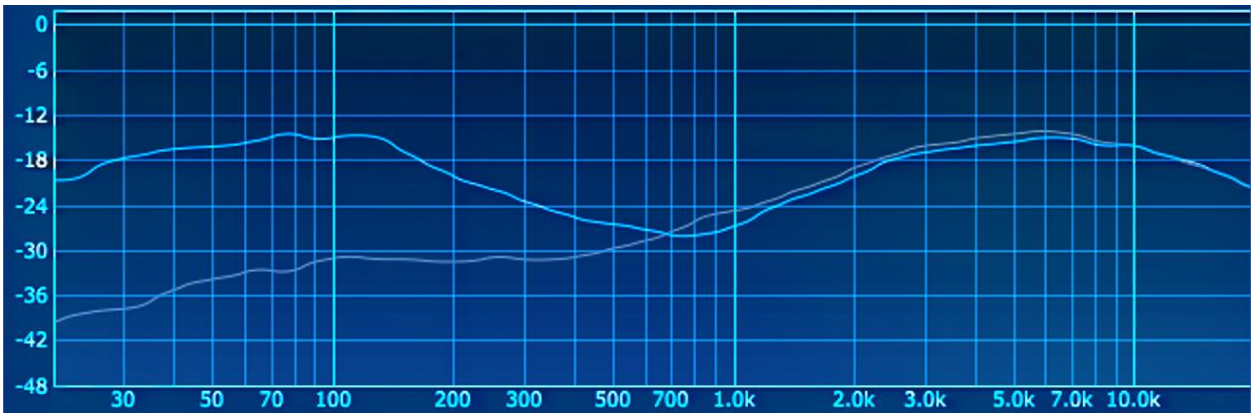
Dynamics: These blocks split the input into High, Mid, and Low bands, using the 3-Band Compressor blocks from the Favorites folder (**3Bd_HiSplitComp.fav**, **3Bd_MidSplitCom.fav**, and **3Bd_LowSplitCom.fav**, respectively).
Pre-Amp EQ: The Parametric EQ adds a narrow cut at 2.5 kHz to sweeten the high-frequency tone.
Amp+Cab: Each amp is tweaked for its specific frequency range by editing the Drive, Sag, and Tone controls. All Cabs use the default miking and cab parameters.
Vol: These three "utility" blocks simplify editing. Enabling a block mutes its respective band. Muting two bands allows tweaking the remaining band in isolation. After editing the preset as desired, you can delete these.
Post-Amp EQ: The pre-Amp EQ creates a sweeter sound by reducing distortion at higher frequencies, but this also reduces the highs. So, the post-Amp EQ boosts the sweeter sound to compensate for not having as many highs going into the amp. The EQ also increases the level, and trims Low Gain by -4.0 dB.
Mixer: The Mid and High bands are panned to 30 left and right, respectively.

PV Vitriol Clean

Peavey amps were never known for being shy, but it's interesting to see that approach applied to a clean amp. It incorporates a natural scoop, and the bass and treble controls can provide a significant amount of boost.

Tip: For a near-flat response, set Bass to 1.0, Mid to 4.0, and Treble to 0.0.

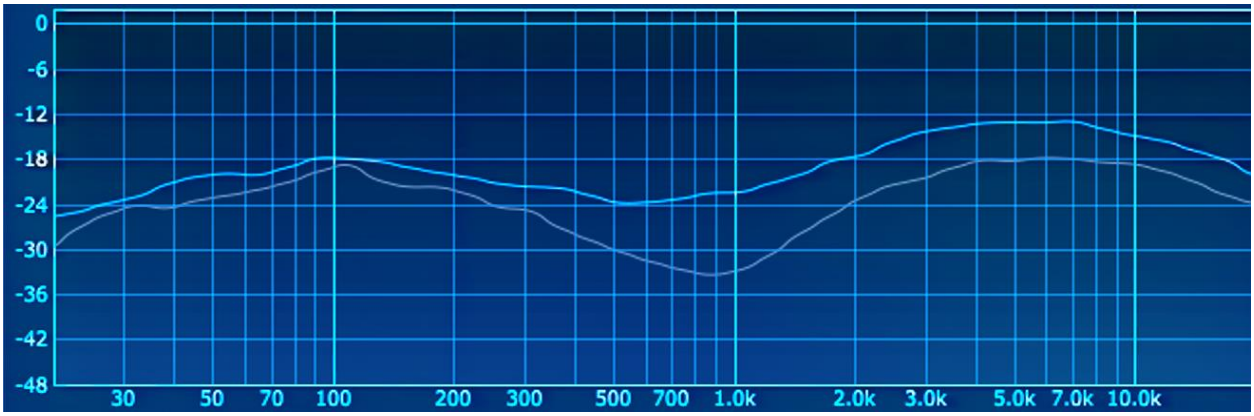
Bass



Boosting gives a major rise in bass response starting around 700 Hz. Cutting starts flattening the response around 500 Hz, which then begins a gentle rolloff at 100 Hz. Whether boosted or cut, the bass response extends well below the guitar's range.

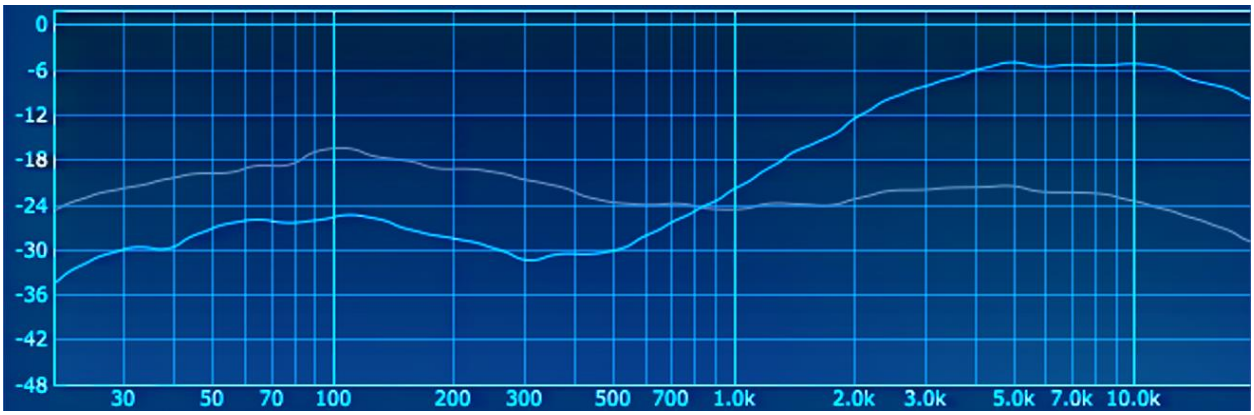
Tip: To keep excessively low frequencies out of the Vitriol Clean, insert a Low and High Cut filter before the amp. Set the Low Cut frequency around 80 to 100 Hz.

Mid



Turning up Mid counteracts the inherent midrange scoop, but doesn't go high enough to cancel it out. Turning down Mid scoops the response more in the 700 Hz – 1 kHz range. Flattening the scoop requires lowering both the bass and treble controls.

Treble



This control affects several different ranges simultaneously, somewhat like the Line 6 original amps. Boosting gives a major treble boost, but also introduces a scoop centered around 300 Hz before climbing back up to a bass bump. When cutting, the response becomes far flatter, while still favoring the bass.

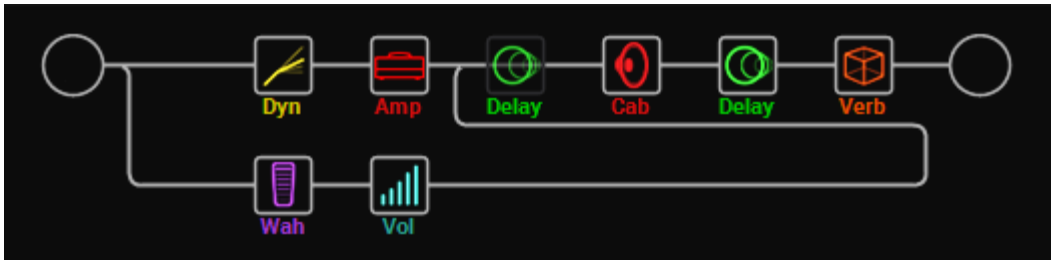
Vitriol Clean Presets

OldSchoolLead

It’s the sound of 1950s slow dances, from the era at the dawn of rock and roll.

Dynamics: To keep strong peaks out of the Vitriol Clean amp, the LA Studio Comp is set to Limit. Aim for Gain Reduction around -9 dB to -12 dB max. With everything bypassed except for Dynamics and Amp+Cab, there should be roughly unity gain through the preset. Bypassing Dynamics gives more distortion. If you want to retain the clean sound with Dynamics bypassed, lower the Amp’s Pre Gain setting.
Amp+Cab: Compared to the default, Pre Gain is 3.1 instead of 5.0, while Treble, Post Gain, Presence, and Depth are set to higher values than the default. The Bias settings are carefully optimized for the Amp. The mic settings are different from the default Cab. A 421 Dynamic subs for the 47 Cond FET. Its position is Center instead of Cap Edge, and the Distance is further away at 5.25".
Mod: It wouldn’t be an old school amp without a Tremolo, right?
Delay: The Cosmos Echo gives just the right amount of vintage tape echo vibe.
Verb: The Legacy ’63 Spring frees up enough DSP for the various other effects and the processor-hungry Vitriol Clean itself.
EQ: The Simple EQ at the end boosts the Mid and adds some brightness.

Clean Lead



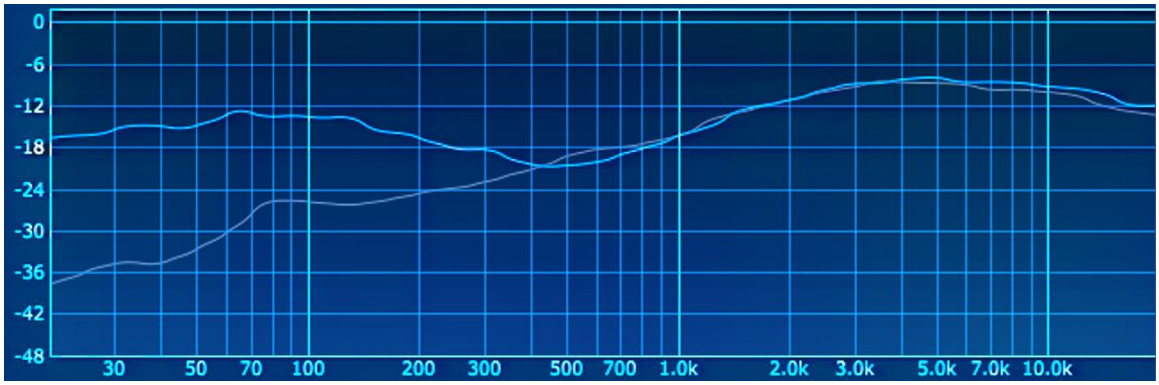
This basic preset has the added expressiveness of a parallel wah effect. The preset loads with the Vibrato (1st delay) and Wah bypassed. Assign the Vol block Bypass to a footswitch so you can bring the wah in or out. As with the previous preset, the Vitriol Clean is sensitive to the input level if you want truly clean sounds.

Dynamics: Please see above. This has the same settings as the **OldSchoolLead** preset.
Amp: Like the above, Pre Gain is 3.1 instead of 5.0. This requires higher Post Gain, Ch Vol, and Master settings than the Amp defaults. Presence, Treble, Depth, and Sag are also higher.
Delay 1: The Mod/Chorus Echo provides vibrato. The Time is 0.0 ms, and the Mix is 100% wet. It's mono, because using the Mod/Chorus Echo's stereo block dilutes the vibrato effect with mono playback systems.
Cab: The Dual Cab combines default 4x12 Cali V30 and 1x12 Open Cast Cabs.
Delay 2 : The stereo Simple Delay creates a wider stereo image. Although it doesn't create phase issues when collapsed to mono, consider bypassing it to avoid the slight slapback echo effect that happens in mono.
Verb: The Dynamic Plate is a good complement to this preset. However, there's enough processing power to allow any stereo reverb, except Nonlinear.
Wah and **Vol:** These two blocks work together. Enabling Vol (the default state) sets its gain to -120 dB, which prevents the wah or dry sound from making it to the output. However, Vol could also alternate between two amounts of gain to bring in more or less of the Wah sound. In any case, the Wah sound is subtle and doesn't overwhelm the preset.

US Dripman Nrm

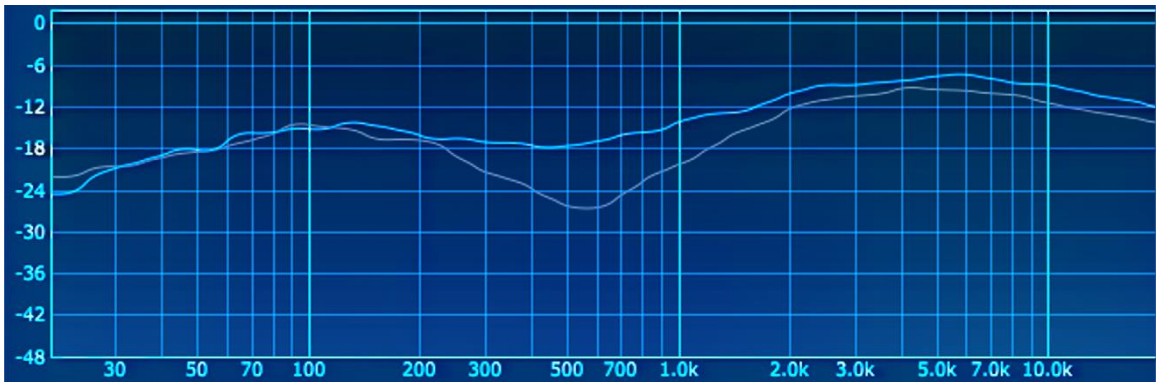
This falls into the “aggressive bass amp that also works with guitar” category.

Bass



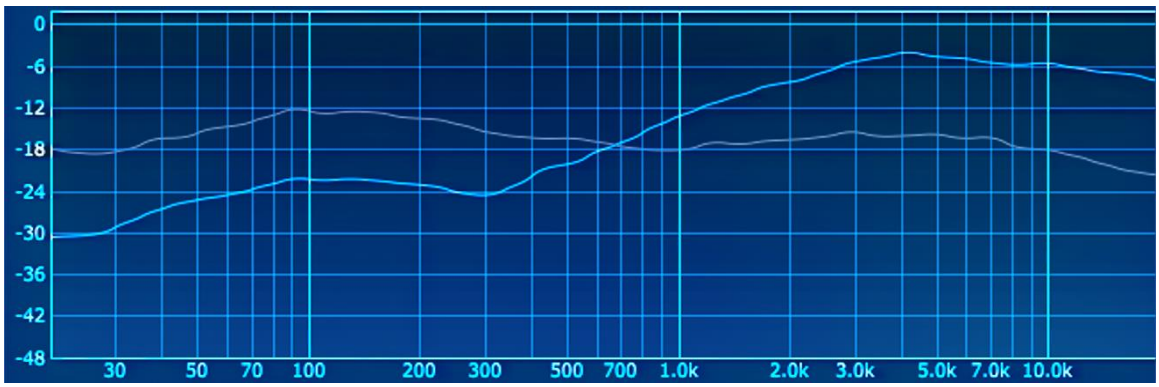
The straightforward bass control gives a broad boost that extends from 400 Hz up to about 60 Hz, where it peaks. Below 60 Hz, the rolloff is almost non-existent. If you play synth bass, or want to use an octave divider on bass, go for it. Cutting rolls off the low frequencies starting at 400 Hz, and slopes downward from there.

Mid



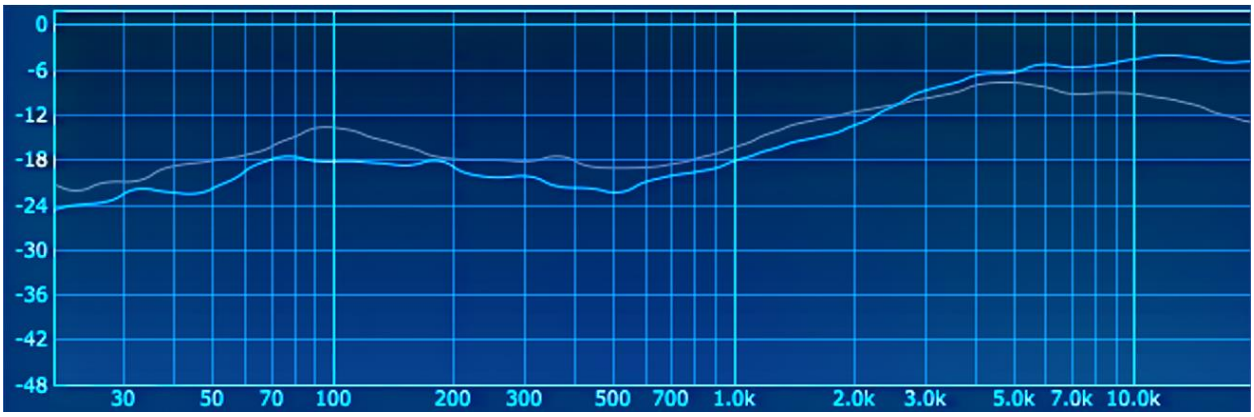
With the tone controls at 5.0 and Bright off, there’s a broad, shallow scoop around 500 Hz. Turning up Mid almost cancels out the scoop. Cutting the Mid control gives a deeper scoop at the same frequency.

Treble



Turning up treble boosts the response starting at 500 Hz, but simultaneously cuts the response somewhat below 500 Hz. Cutting treble reduces the inherent boost that starts around 500 Hz, while adding a gentle emphasis below 500 Hz that peaks slightly around 100 Hz.

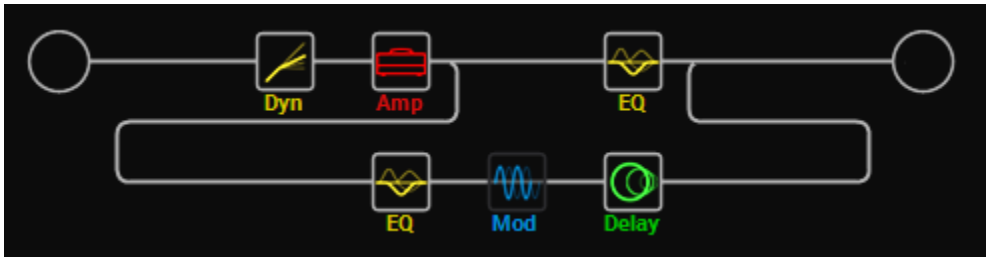
Bright Switch



Turning on Bright counteracts the high-frequency rolloff that normally starts around 5 kHz. This can add “air” because there’s not a whole lot of bass energy up that high, other than the harmonics generated by distortion. Bright also reduces the bass response somewhat.

US Dripman Norm Presets

Drip Bass Stereo



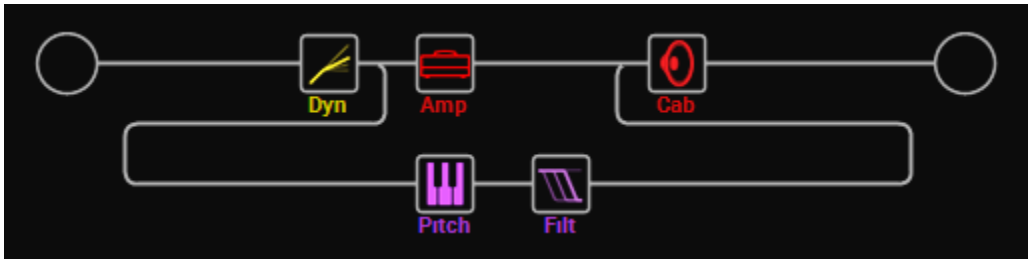
For decades, bass was mono and always panned to center. This was due to the physics of how we hear music (low frequencies are less directional than high frequencies) and the limitations of cutting bass with vinyl. With vinyl, stereo bass could cause the stylus to jump out of the grooves.

Digital audio removes the technical reasons for placing bass in the center. Although I still prefer to pan bass to center, this preset imparts a sense of stereo by splitting the high frequencies off to the left and right.

Dynamics: The Ampeg Opto Comp sounds wonderful with bass. To hear the preset sound as intended, aim for gain reduction in the -7 dB to -14 dB range.
Amp: Changes compared to the default include more Bass, but less Treble, Sag, and the Bright switch is off. For a cleaner bass sound, choose a Drive setting of 2.0 or even lower.
EQ (main path): There’s no cab per se, but this EQ creates an analog cab. The gain for the various bands slopes downward, except for a slight bump at 2 kHz.
EQ (parallel path): The Simple EQ cuts the lows (the main path provides the needed low frequencies) but boosts the mids and highs to feed the stereo image.

Modulation: The preset loads with the Ampeg Liquifier bypassed. Enable it to add a sweet animation to the high frequencies. This complements the lower-frequency gruffness.
Delay: Combining stereo mode with a short delay and zero feedback provides a stereo signal to the mixer's B Path. With a mono output, phase issues are minimal because the level of this path is lower than the main path. However, you still may want to bypass the Delay.
Mixer: The B Level parameter setting is crucial. Too high a setting overwhelms the main path, while too low a setting doesn't provide the intended stereo effect.

Mouthy Drip Bass



Using an envelope filter in parallel with bass is nothing new. It's been the basis of many funk bass sounds. This variation differs by including the 12-string pitch block to give the filter a wider frequency range for processing.

Dynamics: The 3-Band Comp uses the default settings. If it ain't broke…
Amp: Because the parallel path provides a lot of midrange and highs, the amp pulls back the Treble all the way and the Mid a bit, but boosts the Bass to max and turns off the Bright switch. The Drive is also less, but that's a personal preference. Drive is very sensitive to input level. I think the preset sounds best when Drive is set for a light crunch, not heavy distortion.
Cab: The settings are the same as the default, but with more level.
12-String: You probably wouldn't expect to see this Poly block in a bass preset, but it creates a harmonically complex signal. This gives the Autofilter more interesting audio to process.
Autofilter: I didn't want this to sound like a caricature of a funk sound. So, the relatively long Attack and Decay give a less abrupt and less percussive effect. But this block also has the most potential for personalization. Play around with the settings until you obtain the kind of sound *you* prefer.

Mandarin 200

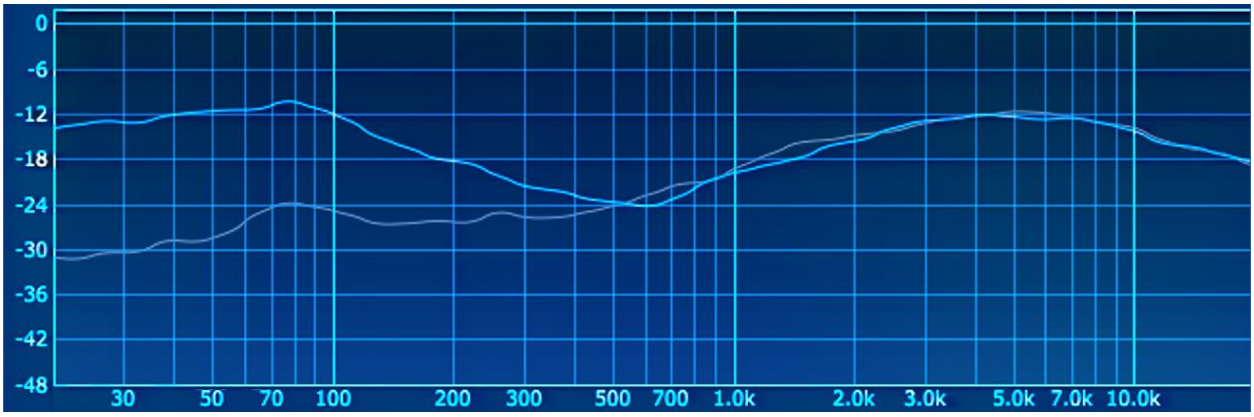
With the tone controls at 5.0, the response has a broad, shallow scoop and some high-frequency emphasis. This amp can give full, round tones, as well as throw in some harmonics for “growl.”

Drive is flexible. It can give clean sounds, but you can crank it up for heavy distortion.

Tip: Try following the Amp with the 10 Band Graphic instead of a cab. This lets the full bass sound through and gives interesting options for tailoring the midrange.

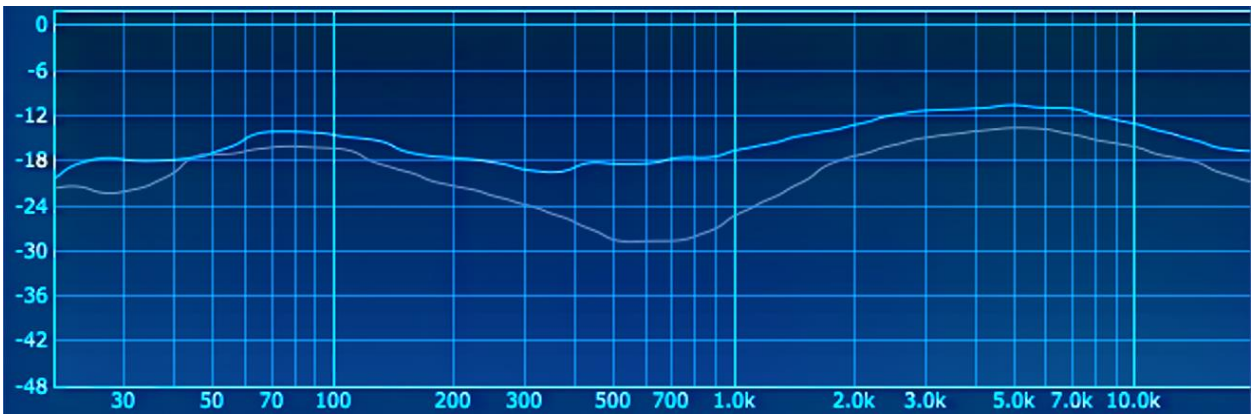
Tip: For a fuller, rounder bass sound with the default 6x10 Cali Power cab, swap out the default 57 Dynamic mic for the 4038 Ribbon mic. Choose 3.5 for position, and 3.00" for Distance.

Bass



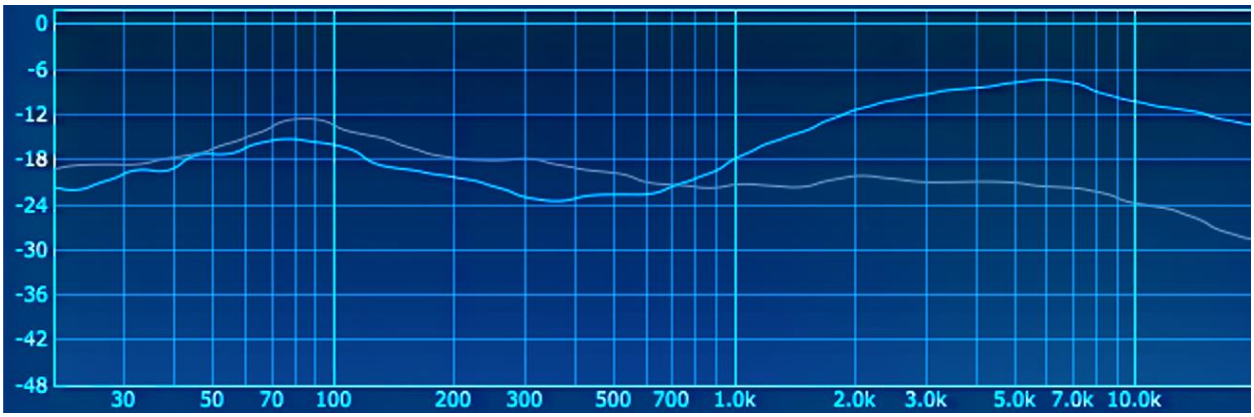
Turning up Bass drops the scoop slightly deeper, while boosting frequencies below around 500 Hz. Cutting bass flattens out the response somewhat from 500 Hz on down.

Middle



Boosting Middle makes the scoop a bit shallower, but cutting increases the scoop depth considerably. Cutting also de-emphasizes the high and low responses somewhat.

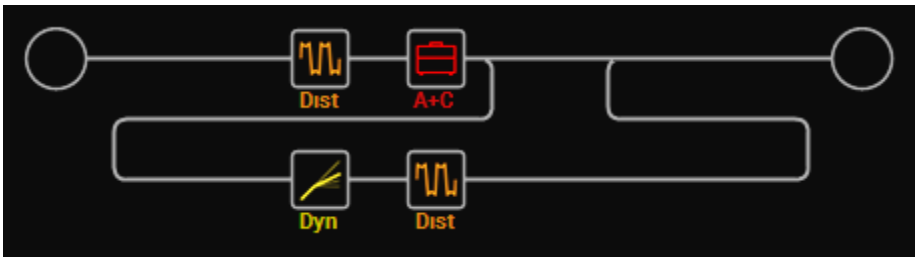
Treble



Boosting pours on the treble above 500 Hz. It also makes the scoop a little deeper and drops the bass response slightly. Cutting tilts the treble response downward, and the bass response upward.

Mandarin Bass 200 Presets

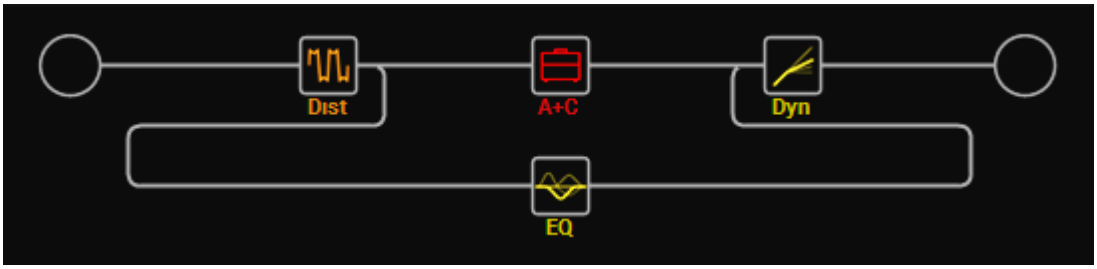
MandrinBriteAtak



This preset adds a trace of prog rock treble to the otherwise big sound of the Mandarin 200 Bass.

Distortion (main path): Compared to the default settings, the Regal Bass DI hypes the sound going to the Amp+Cab by boosting the Treble and increasing the Volume slightly. Dialing back the Bass emphasizes this preset's brighter nature.
Amp+Cab: There's more Bass, Treble, and Sag than the default amp settings. I prefer a slightly higher Drive setting, but that's a matter of taste. For a smoother sound, the Cab swaps out the 57 Dynamic mic for a 414 Condenser mic.
Dynamics: This has the uncommon task of compressing the Amp+Cab *output* before passing it along to another stage in a parallel path. With the LA Studio Comp, aim for gain reduction in the -6 dB to -9 dB range.
Distortion (parallel path): The Regal Bass DI (by the way, don't overlook this for guitar) uses the maximum Treble setting. This contributes to the bright attack sound.

MandrinBassBoost



Now that the **MandrinBriteAtak** provided a preset for emphasizing treble, this one goes in the other direction and emphasizes a seismic bass sound.

Distortion: In the Regal Bass DI, Treble pulls back all the way to 0.0. Bass remains the same as the default.
Amp+Cab: Compared to the default, Mid is down a little bit, and Treble is all the way down. Sag is at minimum to retain a more percussive quality.
EQ: It's all about the bass. The 10 Band Graphic turns down all the faders above 250 Hz. The parallel placement allows adding the pre-Amp bass sound back in with the post-Amp bass sound.
Dynamics: The LA Studio Comp processes the mix of the amp sound and the boosted bass. Aim for gain reduction in the -6 dB to -9 dB range.

Amp Blocks in Version 3.80, with Presets (1.5)

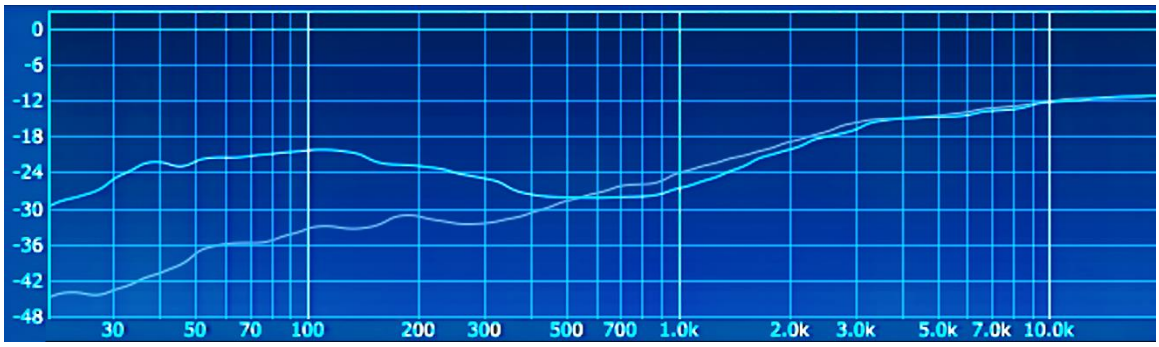
V3.80 introduced six emulations of classic guitar amps, and four new cabs. The *Free Files folder > Chapter 2 Presets > 3.80 Amp Presets* folder includes new, custom presets for these amps. The Chapter 2 Presets folder also has a setlist, **eBook v1.5 New Presets.hls**, with the new presets in this version of the eBook.

As with the descriptions of the 3.70 amps, the tone control response graphs show the control's response over its maximum and minimum settings. These graphs do not include the Amp's Cab. They relate solely to the Amp. The curves set all controls to 5.0 except for the control being analyzed. Each control shows two curves. The brighter line is the maximum setting (10.0), while the darker line is the minimum setting (0.0).

US Super Nrm (1.5)

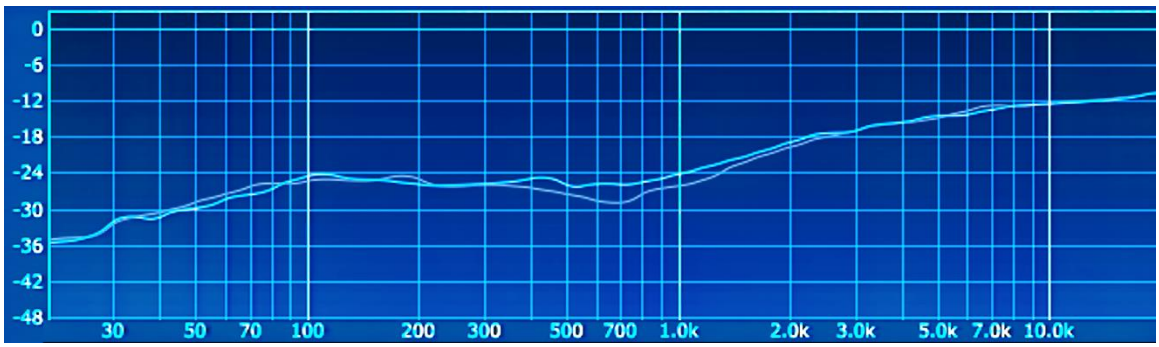
The Fender Super Reverb was the amp I lusted after in my formative years of playing live. The bright, present sound was ideal for pop, rock, and surf music. Line 6 has nailed the Super Reverb's soul. The tone control curves show why this combo amp could cut through mixes, as well as stand out in live performance.

Bass



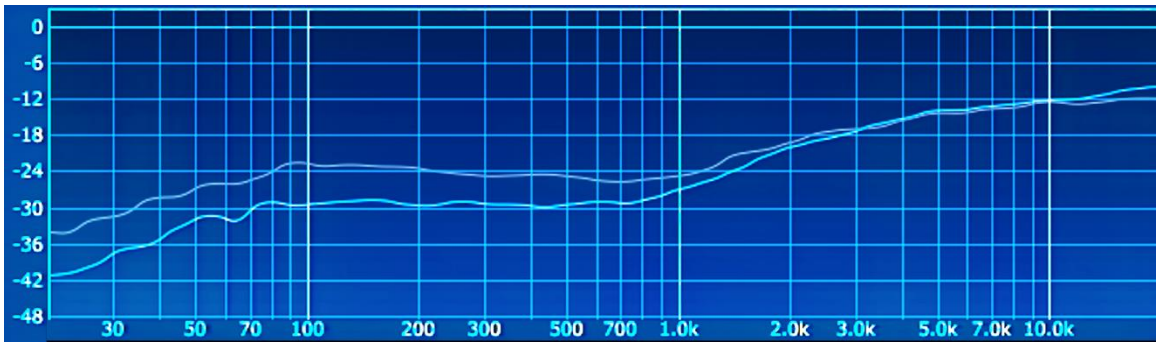
Turning up Bass adds a broad boost around 100 Hz. Because of the bright treble response, turning up Bass to maximum coincidentally creates a midrange scoop around 600 Hz. When fully cut, the frequency response curve is like a Tilt EQ, where the response follows a linear downward slope from high to low frequencies.

Mid



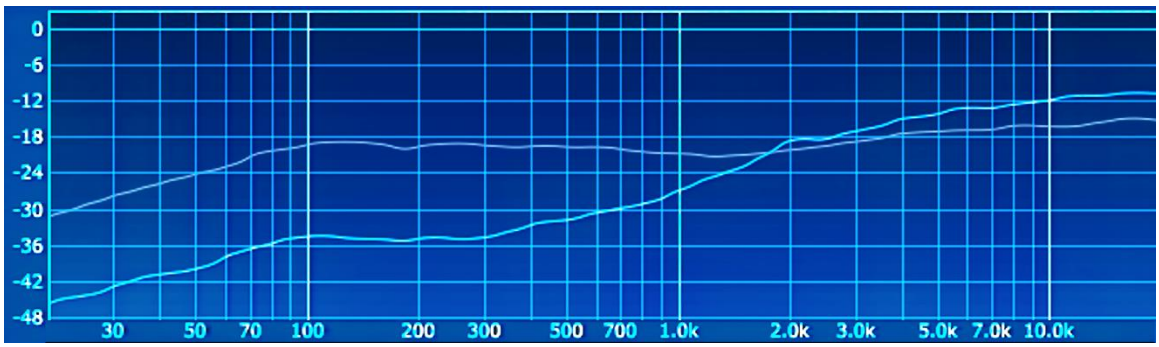
The Mid control has a subtle influence. It affects the frequency range from 400 Hz to around 2 kHz. Turning up Mid gives a slightly more "forward" sound, while the minimum setting places the guitar a little further back in a band or mix setting. (Note: the original Super Reverb didn't have a Mid control. It was added in 1978.)

Presence



Presence usually means an upper midrange or treble boost. However, turning up Presence mostly reduces frequencies in the range from bass to mids. This emphasizes the highs to give more presence, but thins out the overall sound somewhat.

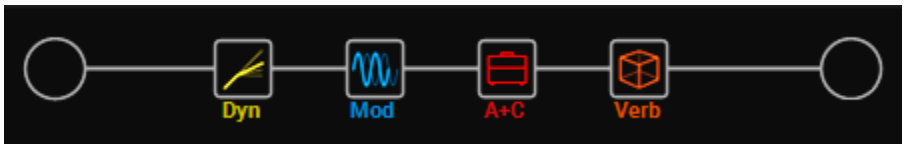
Treble



The Super Reverb is an inherently bright amp. Boosting Treble adds some brightness but more significantly, it also attenuates the lower mids and bass to make the treble seem louder by comparison. Turning treble down all the way, with the other tone controls at 5.0, gives an essentially flat response. **Tip:** Assign this control to a foot pedal to vary between a warmer, quieter rhythm sound and a brasher, brighter lead sound.

US Super Nrm Presets

Fendemaggie



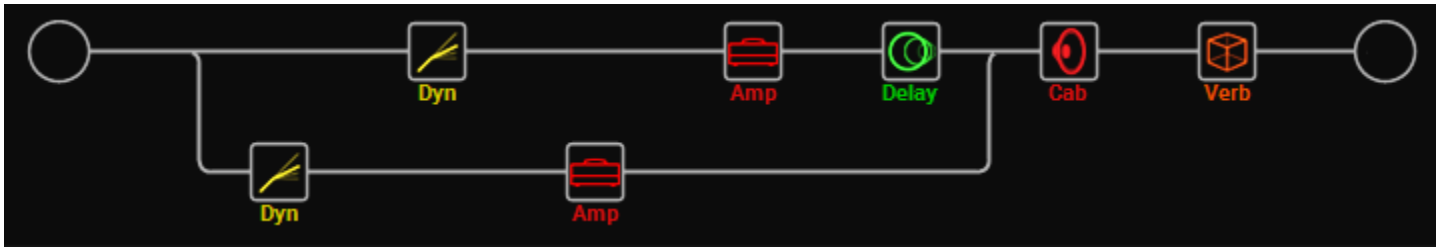
This preset is inspired by the Magnatone amps that differentiated their brand by using vibrato instead of tremolo. (Yes, Fender called tremolo “vibrato.” We’ll let that slide.)

Dynamics: LA Studio Comp applies mild compression. Adjust Peak Reduction to taste. To apply compression equally to high and low frequencies, Emphasis is turned down.
Mod: Although several effect blocks can produce vibrato, I like using the Gray Flanger. Setting its Mix parameter to 100% gives delayed sound only. Modulating it produces vibrato. Width determines the vibrato depth. A touch of Regen emphasizes the effect.

Amp+Cab: This preset tweaks the default tone control settings for more mids and less treble than the Fender Super Reverb. Also, the Cab uses a 160 ribbon for a warmer sound.
Verb: The default Hot Springs settings provide a vintage vibe.

Twin Guitars

This unusual stereo preset makes it sound like two guitarists are playing in the left and right channels. Listen over headphones or stereo speakers. It's uncanny.

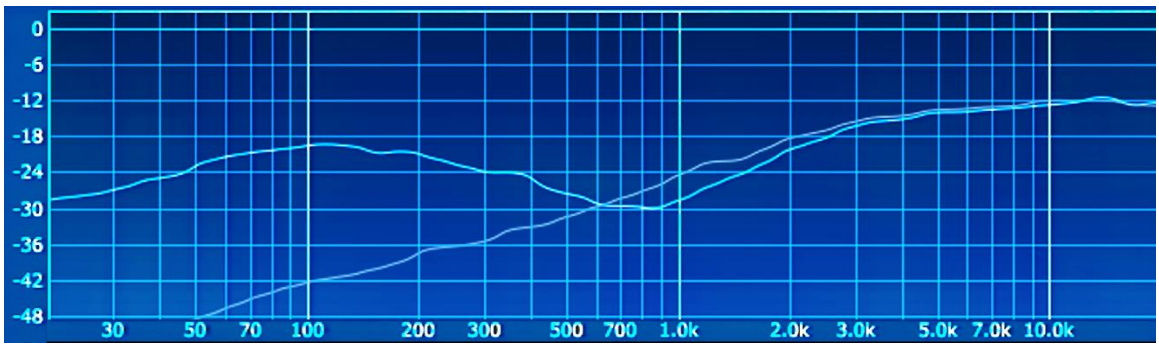


Dynamics: The 3-Band Dynamics processors are not acting as dynamics processors. Instead, the one in the A path passes frequencies below 350 Hz *and* above 1.0 kHz. The one in the B path passes frequencies between 350 Hz and 1.0 kHz. So, the parallel guitars have different tonal qualities.
Amp: The Amps are the Super Nrm amps with their default settings. The settings are just a starting point. To change the sound of the "two" guitars, tweak the tones of the two amps.
Delay: To further differentiate the sounds of the two guitar paths, the Mod/Chorus Echo adds 24 ms of delay with subtle frequency modulation.
Cab: The Dual Cab maintains the stereo image. Both Cabs use the same default 4x10 US Super settings, with two exceptions. The Cabs are panned hard left and hard right, and the Path B Cab's level is a little higher to match the Path A level.

US Super Vib (1.5)

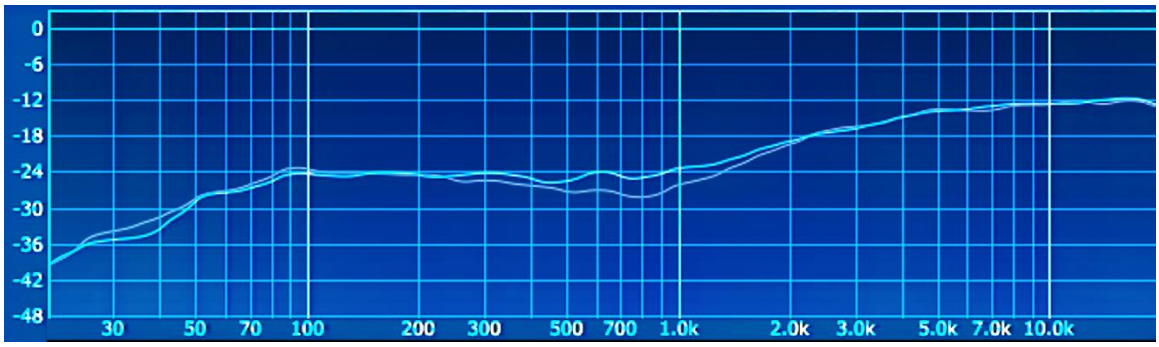
The vibrato channel has an extra gain stage. When pushed, this adds some extra grit compared to the Nrm channel. It also seems inherently brighter, and subjectively, somewhat more dynamic. Use this channel if the US Super Nrm channel is cleaner than you'd like.

Bass



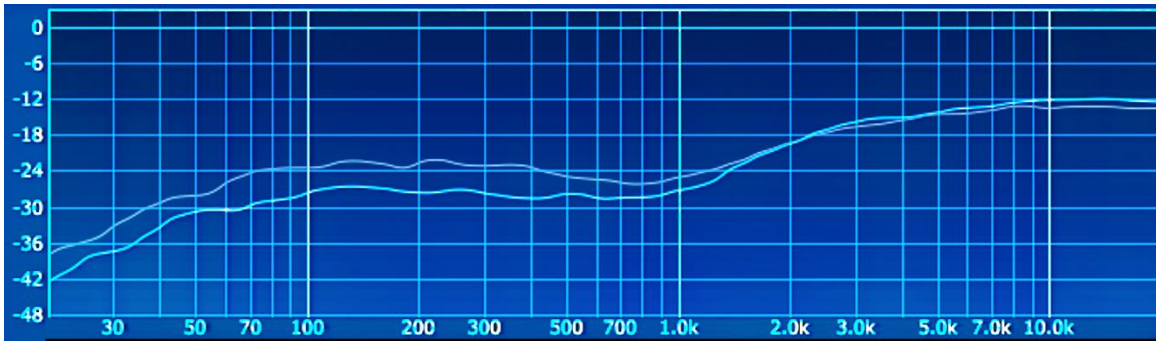
The boost response is much like the Nrm channel. However, cutting rolls off more quickly under 1 kHz, and attenuates low frequencies more drastically.

Mid



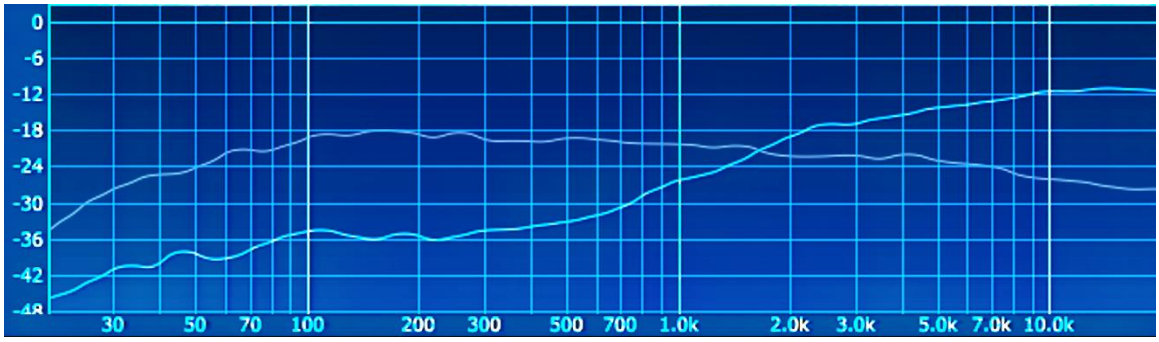
The mid response is essentially identical to the Nrm channel. Perhaps Fender simply “ported over” the existing Mid control circuitry.

Presence



Again, there’s little response difference between the Vib and Nrm channels.

Treble



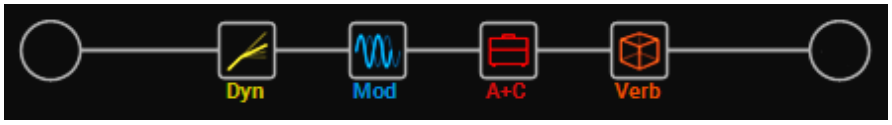
The boost response is similar to the Nrm, but cutting differs. The Nrm channel tends to flatten out the high-frequency response above 1 kHz. The Vib channel rolls off highs above 1kHz.

US Super Vib Presets

Surfism

The surf guitar sound starts with your guitar and playing technique. Ideally, your guitar will have:

- Solid body construction
- Single-coil pickups
- A whammy bar (aka vibrato tailpiece)
- At least .010 gauge strings (preferably a little heavier), and a hard pick to go with the heavy strings.

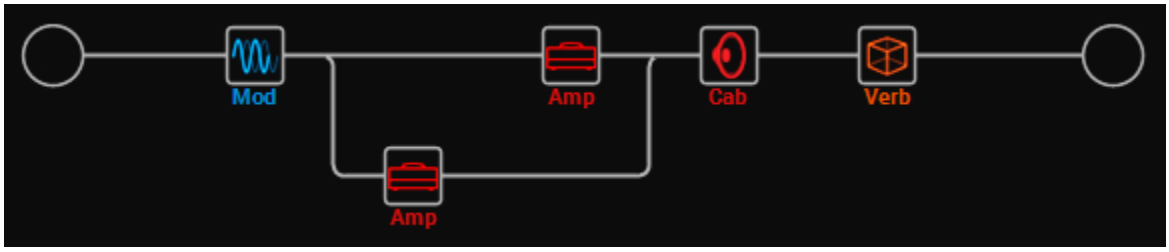


Surf music wants a clean amp with some grit, lots of treble, and spring reverb. Tremolo is optional. To increase sustain while using the whammy bar, insert a compressor before the amp to bring up string levels as they decay.

Dynamics: Adjust LA Studio Comp's Peak Reduction parameter for the desired amount of sustain.
Mod: The 60s Bias Trem provides a satisfying tremolo effect. Placing it before the Amp+Cab emulates the pre-power amp position of traditional tremolo designs.
Amp+Cab: Turning on the Bright switch while reducing the Treble gives a bright sound, yet avoids excessive harshness. The Cab substitutes a 57 Dynamic mic for the default 47 Cond FET.
Verb: The Double Tank uses its default settings.

CabStackStereo

The Dual Cab gives a stereo image. This preset is excellent for recording or loading into Helix Native.



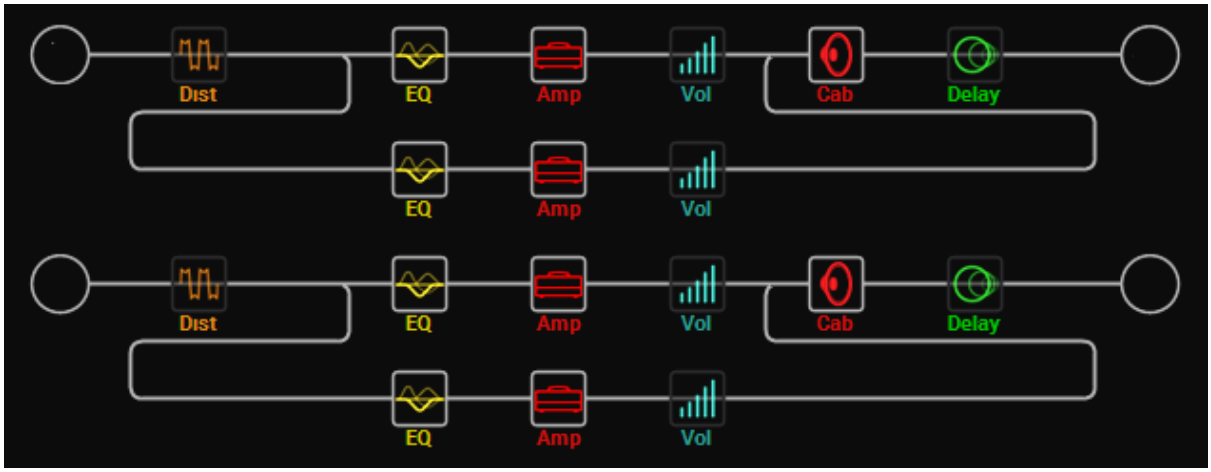
Mod: To add a variation, the Harmonic Tremolo substitutes for a standard tremolo effect.
Amp: Both amps use the default settings. This preset highlights how it's possible to change the sound solely by "stacking" two cabs for a stereo routing and using different mics, rather than changing Amp settings.
Cab: The Dual Cab uses two 4x10 US Super cabs. One pans full left, the other pans full right. The mixer pans the A and B paths oppositely to maintain the stereo image created by the cabs.
Verb: The stereo Hot Springs uses its default settings.

CabStackStereo2

This is the same as CabStackStereo, except that the Dual Cab uses different speakers (4x10 US Super and 4x10 Tweed P10R) and different mics (57 Dynamic and 906 Dynamic). Making these two simple changes alters the sound considerably compared to the CabStackStereo preset. The result is a brighter, more aggressive sound.

4B.DirtySuprVib

This multiband preset requires a four-path Helix (e.g., Helix Floor or Native). It pushes the amps hard to bring crunch to the US Super Vibe. However, splitting the audio into four bands reduces intermodulation distortion within the individual bands. So, despite the "dirt" that crunch adds, the US Super Vibe still has a clean sound that articulates well. This preset also has an outstanding stereo image.



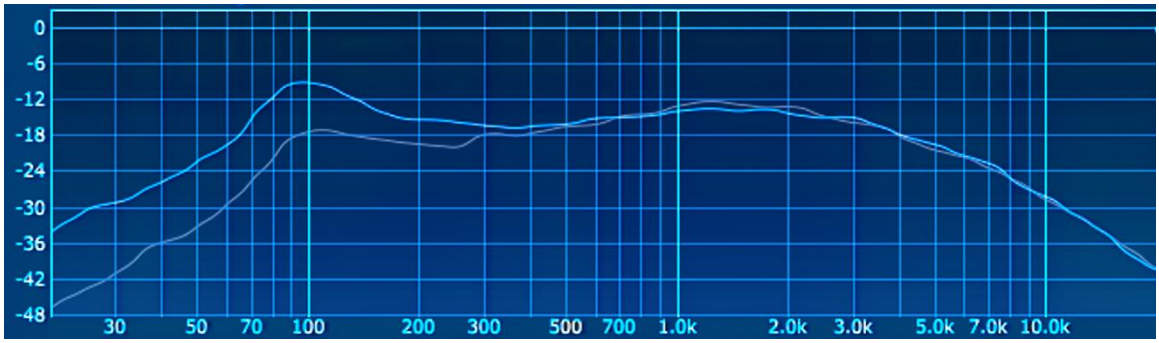
Distortion: There's enough processing power left over, even with four amps and two dual cabs, to accommodate distortion. The preset uses the Heir Apparent, but any distortion will work. You can also try different distortions for Path 1 (lows and low mids) and Path 2 (high mids and highs). Or, distort one pair of bands but not the other pair.
EQ: The 10-band graphic EQ blocks separate the audio into different frequency bands.
Amps: These have higher Drive settings than the defaults and edited Bias parameters.
Vol: The Vol clocks are for diagnostic purposes. Enabling a Vol block mutes its path. This allows isolating specific bands for further editing. After finalizing any edits, you can delete them.
Dual Cabs: The mics and mic positions are quite different from the default, because they're optimized for the individual frequency bands.
Dual Delays: These emulate a room's reflections. (Allocating processing power for Distortion blocks means there's not enough power left over to add stereo Reverb.) Bypass both delays when using a mono amp in a room, because adding synthetic reflections on top of natural reflections rarely works well.

The preset opens with the delays bypassed. For recording, enable the Dual Delays to enhance the stereo image.
Tip: To have several stereo Reverbs available, delete the distortion blocks.

German Xtra Blue (1.5)

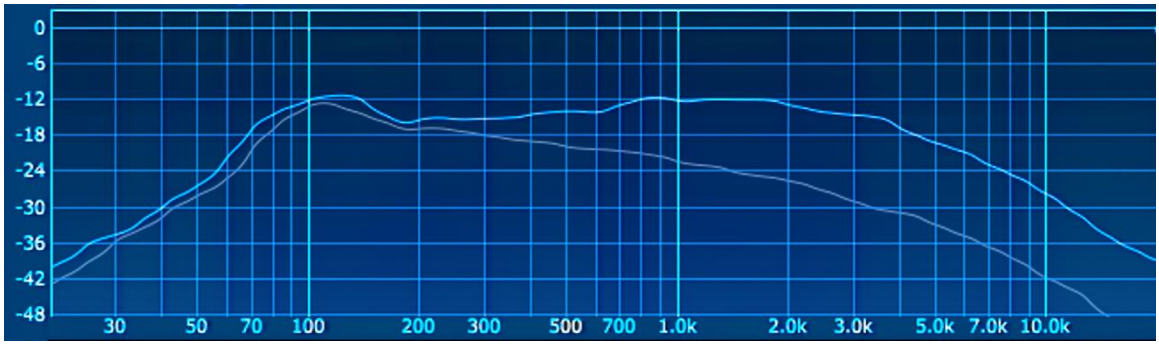
The red and blue color scheme makes sense. The high-gain red channel has an aggressive character. The blue channel has more of an overdrive, crunchy feel. Blue works well for rock, modern country, and (whether by accident or design!) blues. It also has a chunky, beefy midrange sound which the Red doesn't have. The tone controls have a very different design compared to the Red.

Bass



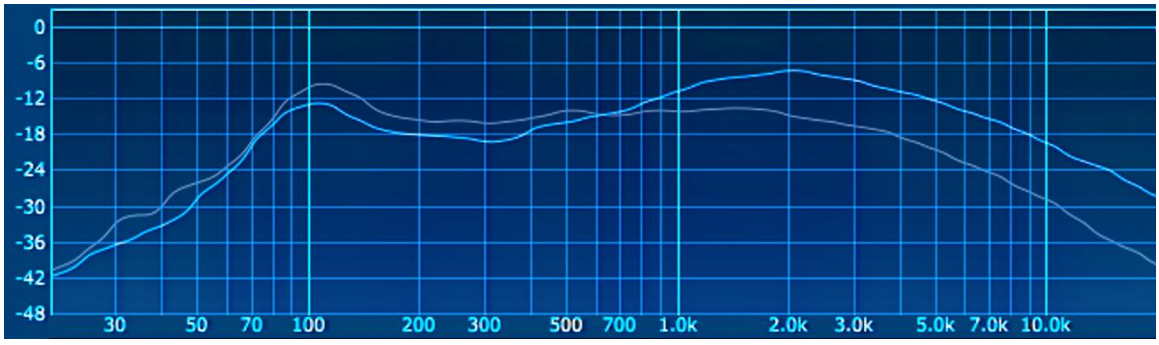
The bass boost concentrates most of its energy around 100 Hz. The cut starts rolling off around 500 Hz, but the slope gets steeper past 100 Hz. This graph also highlights the inherent broad midrange emphasis centered around 1.5 kHz. Unlike the Red, the Blue puts more emphasis on warmer sounds.

Mid



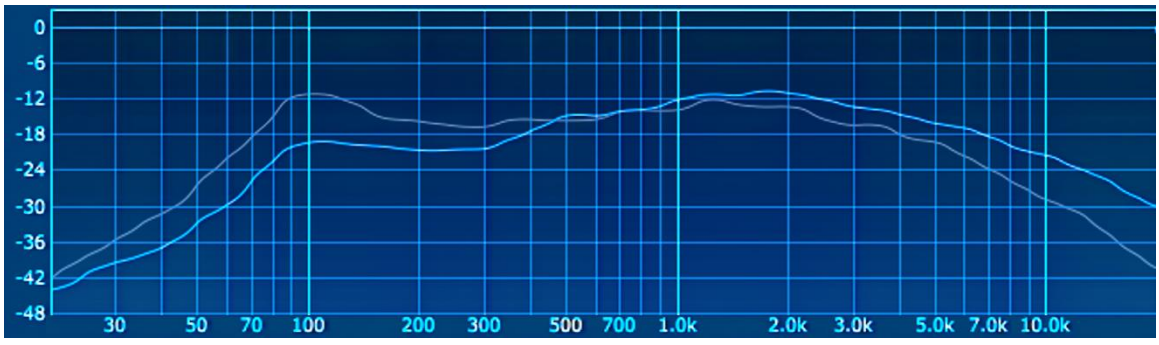
The Mid starts boosting at around 200 Hz and continues into the upper midrange. Meanwhile, when cutting, mid and high-frequency attenuation starts on an almost-linear downward slope at 200 Hz. This may not seem particularly useful. However, boosting the bass and treble, while cutting the Mid control, creates a scoop.

Treble



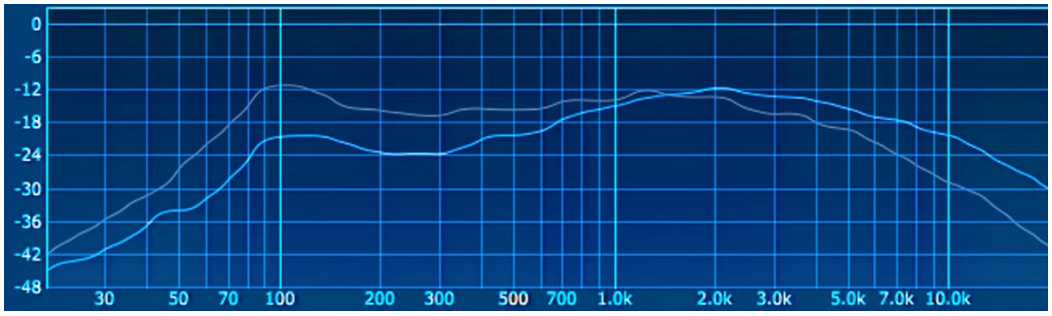
Boosting gives a major treble increase in the upper mids and highs, yet reduces the bass bump around 100 Hz. Cutting leaves the lower midrange and bass bump at 100 Hz intact but starts rolling off highs around 1 kHz.

Bright 1



This graph compares the difference between Bright 1 On (brighter line) and Off. When On, it adds the expected high-frequency boost but to increase the apparent brightness, also starts reducing low frequencies starting around 500 Hz. The bass bump is also flatter.

Bright 2



Bright 2 On increases treble similarly to Bright 1 but reduces the low frequencies more (particularly in the 200 to 500 Hz range). This increases the subjective sense of brightness.

Boost, Structure, Depth, Style, and Class

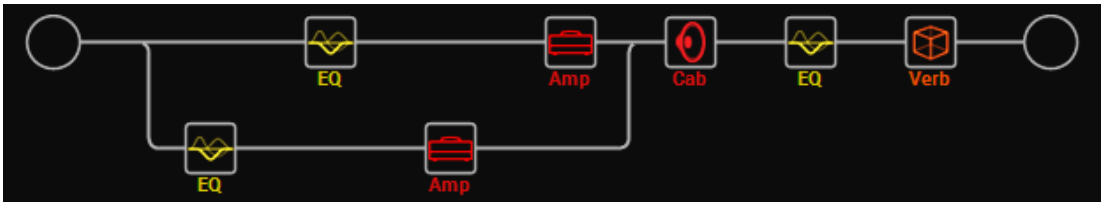
These parameters interact, so there are many possible sound combinations.

Boost: This increases gain to give a more aggressive sound.
Structure: The Low setting raises the gain level for where saturation begins. This favors cleaner sounds. High lowers the gain level for where saturation begins. This favors grittier sounds.
Depth: Tight has the least amount of low-end resonance. Medium increases bass but also increases the overall level. Loose gives the most bass. It increases overall level compared to Tight, but not as much as Medium.
Style: For a more vintage feel, Triode gives a warmer tone with less headroom. Pentode is well-suited to metal and heavy rock. The Pentode sound is open and punchy, with more headroom.
Class: This emulates the tube's mode of operation. Class A is harmonically richer, for a more singing tone. Due to its inefficiency, it was more common in lower-power amps (e.g., Vox AC30). Class AB has more headroom and is louder, but also introduces faint amounts of crossover distortion into the audio.

German Xtra Blue Presets

SweetCrnchRhythm

This preset gives a satisfying, dynamically responsive "playing in a room" crunch sound. It works great for live performance using a mono amp. In the studio, the sound opens up into a full-sounding stereo image.

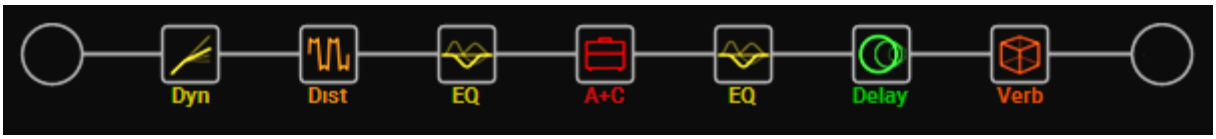


Pre-Amp EQs: One 10-Band Graphic EQ cuts the 31.25 Hz, 125 Hz, 500 Hz, 2 kHz, and 8 kHz bands. The other EQ cuts the other bands. This creates a stereo image that collapses well to mono.
Amps: These use the default settings.
Dual Cab: The default 4x12 Uber V30 Cabs have custom mikings. The left channel uses a 906 dynamic mic. The right channel uses a 30 dynamic mic. They're positioned for a sweet character.
Post-Cab EQ: For more "weight," turn up the Simple EQ's Low control. To push the guitar more forward, increase Middle (its Frequency is 900 Hz, but change as needed). To increase brightness, turn up Treble.
Verb: The stereo Dynamic Ambience reverb wraps a room sound around the guitar.

There's no processing power left with these blocks in place. However, for live performance, you likely won't need the Dynamic Ambience reverb. Removing it frees up enough power to insert most Distortion blocks at the chain's beginning, as well as a wah. Or, do post processing with mono or stereo delay and/or modulation.

Creamy Lead

The additional processors in this versatile preset give different sustain and distortion characteristics, depending on which effects you choose to enable or bypass. Also consider editing the effect parameters.

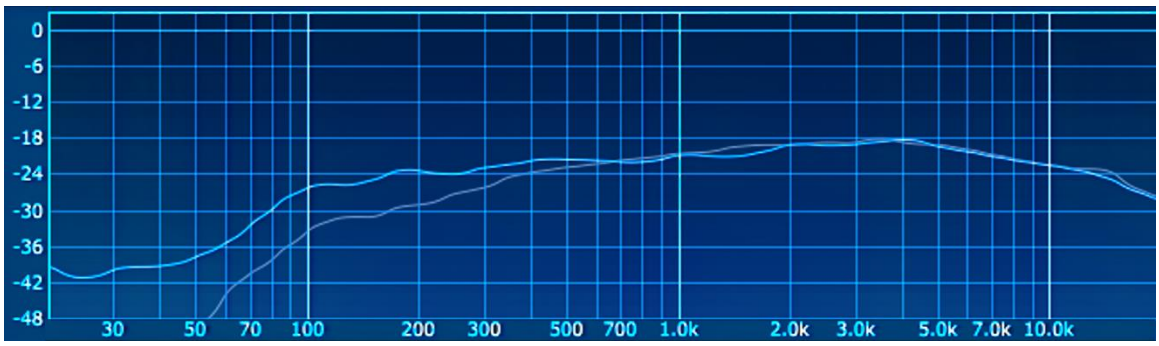


Pre- and Post-Amp+Cab EQs: These create the creamy character. The pre-Amp+Cab EQ cuts the response at 1 kHz. The post-Amp+Cab EQ increases treble to restore brightness, but the sound is creamy due to there being less high-frequency distortion. So, the upper midrange adds brightness, rather than the treble frequencies.
Amp+Cab: The settings are close to the defaults. However, the Bias is tweaked, there's more treble, and Sag is lower to give more attack on leads.
Dyn: The Rochester Comp's purpose is to increase sustain when not using distortion. The basic preset responds to dynamics, but for leads, you can prioritize sustain over dynamic response by enabling the Dyn block.
Dist: The Heir Apparent kicks in distortion for a nastier, dirtier lead sound. Use it with or without the Rochester Comp dynamics, depending on how much sustain you want.
Delay: There's enough processing power to include any one of several stereo delays. I chose the Adriatic Delay because it can give a smooth background echo that stays out of the way of the main lead.
Verb: Similarly to the Delay, you can choose among several stereo reverbs. For a vintage feel, the Hot Springs reverb works well. For a more refined character, choose the Hall or Plate reverb. Or, go for a more unconventional reverb. I like the way Glitz or Searchlights sound with this kind of lead.

German Xtra Red (1.5)

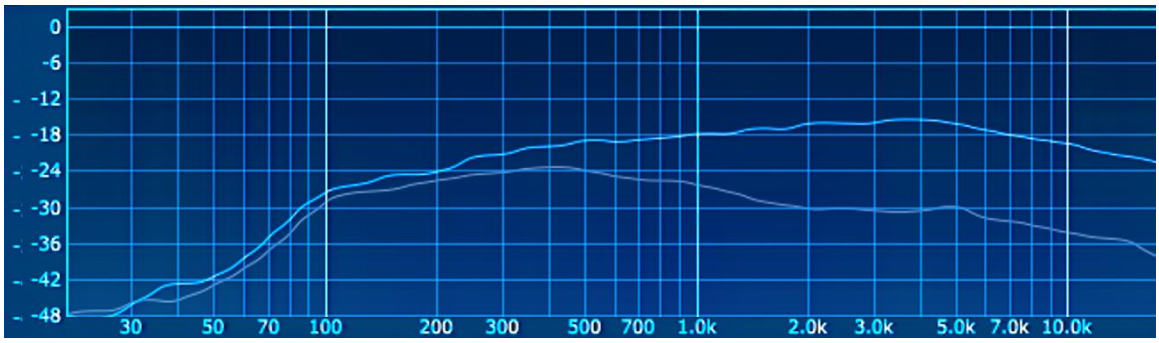
Based on the Bogner Ecstasy 101B, the red channel is designed for high-gain, metal/hard rock-friendly sounds. The heavier saturation lends itself well to more aggressive applications. The low end is tight, and there's a midrange response I'd call "enthusiastic" (that's a compliment).

Bass



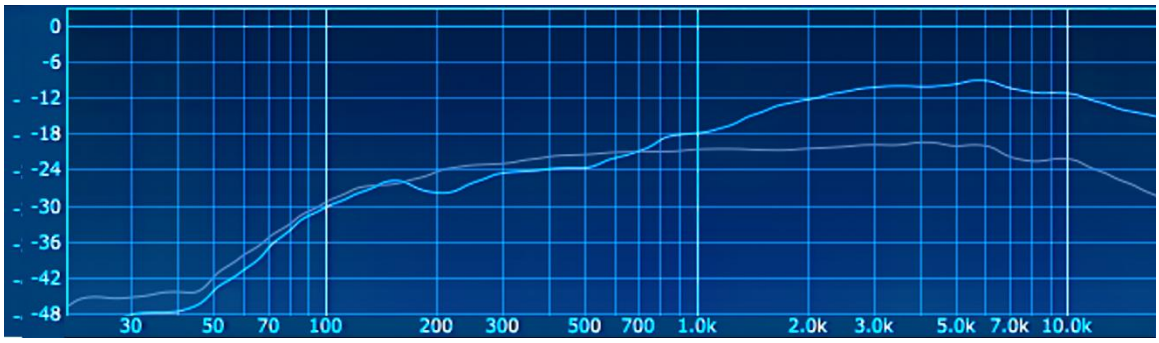
Instead of taking the Blue channel's approach of adding a peak, boosting the bass is more about flattening the response down to 100 Hz. This gives a fuller low end. Cutting rolls off the response in the bass range. As a result, the overall tone is thinner.

Mid



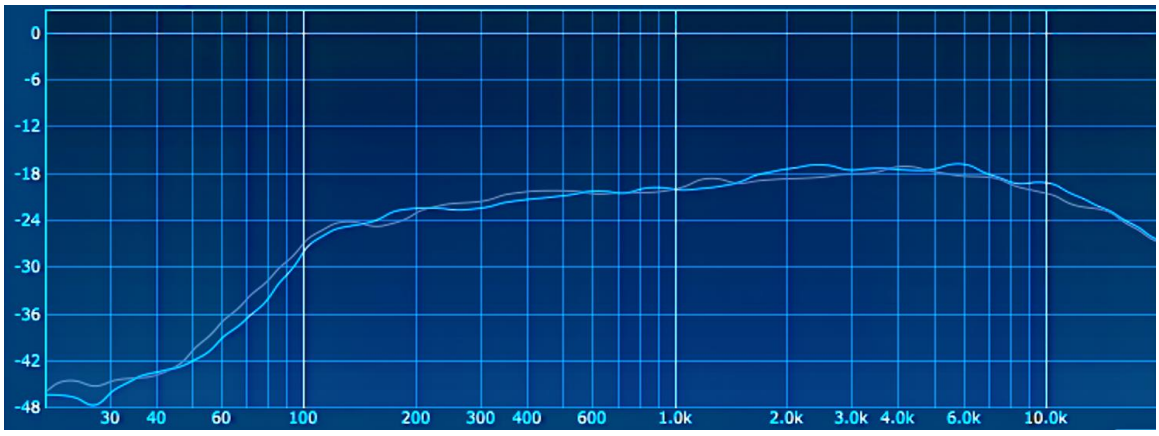
Mid boost increases response starting in the midrange, but this boost even extends into the treble region. Cutting creates a linear downward slope starting at 400 Hz that attenuates high frequencies. As with the Blue channel, to create a midrange scoop boost the Bass and Treble parameters while cutting the Mid control.

Treble



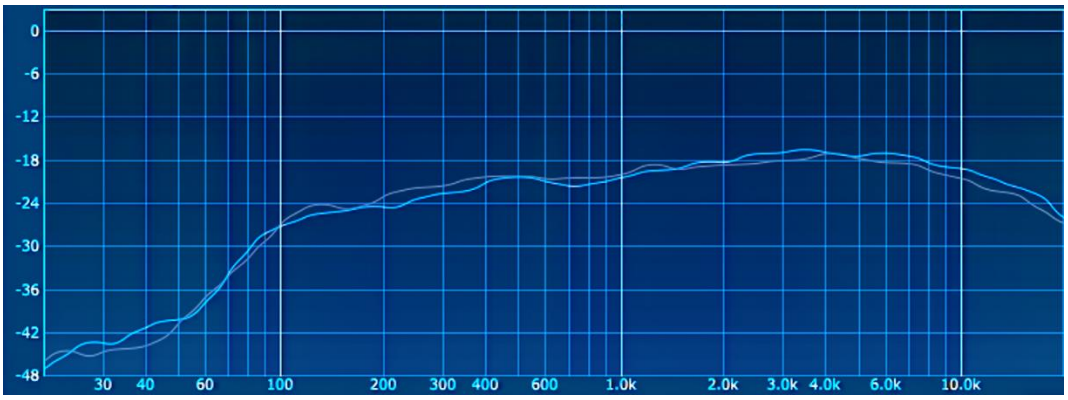
Boosting provides a pronounced response increase in the upper mids that reaches its peak around 6 kHz. Above that frequency, the boost tapers off gently. Cutting Treble flattens out the high frequency response. This helps avoid harshness with bright sounds. Below 1 kHz, boosting or cutting have mostly the same effect on the sound.

Bright 1



Unlike the Blue, the Bright switches have a subtle effect. This is probably for the best, given the brightness induced by the distortion. Enabling Bright 1 reduces low frequencies somewhat below 150 Hz. The difference is slight, but audible. I double-checked these graphs just to make sure they reflected what I was hearing.

Bright 2



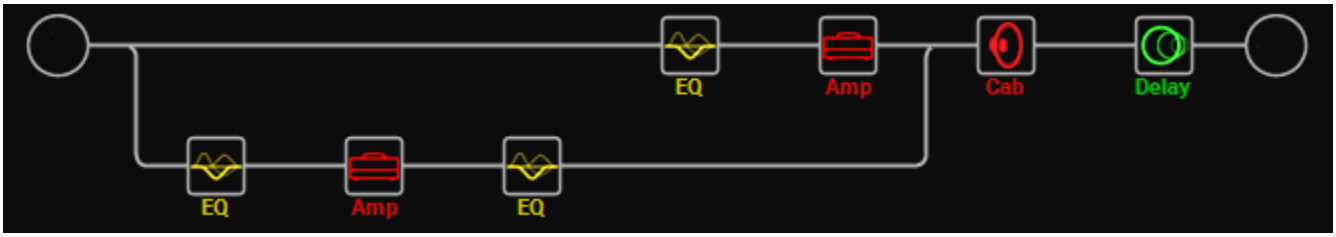
This gives a slightly more even treble increase compared to Bright 1, but it's still super-subtle. Enabling Bright 2 also reduces the lower midrange response slightly.

Boost, Structure, Depth, Style, and Class

These provide the same functions as described above for German Xtra Blue.

German Xtra Red Presets

BiAmpGrmnXtraRed

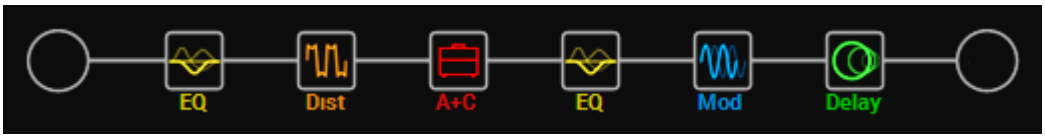


Bi-amping can give a new character to a familiar amp. In this case, the sound is smoother and more focused, yet still aggressive. Compare it to the German Xtra Red Amp+Cab by itself, and you'll hear an obvious difference. Although this preset works well in mono for live performance, it shines in stereo because the Delay expands the overall image.

Split Crossover: This is in bi-amp mode. The crossover frequency is 560 Hz.
EQ: The pre-Amp EQs in both paths tune out specific frequencies with the goal of creating a smoother, more defined sound. The post-amp EQ in Path B smooths the sound further. When you bypass an EQ, it's easy to hear how it influences the sound.
Cab: This Dual Cab uses two 4x12 Uber V30 cabs to be faithful to the original Amp+Cab assignment. However, the miking is optimized to shape the sound of the high and low frequency bands.
Delay: Although the two speakers in the Dual Cab are panned slightly left and right, bi-amping rarely lends itself to full stereo. The Delay provides a broad stereo image that collapses well to mono.

Red Leader

This preset is designed for leads. *Some parameters will* require *editing to tame this preset for your needs.* This is due to the amp's high-gain nature, as well as potential interaction among the various controls. The preset opens with the post-Amp+Cab EQ and Distortion blocks bypassed.

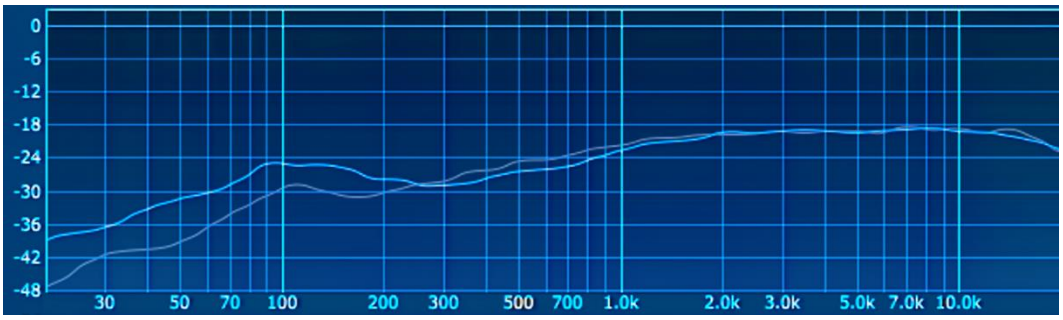


Input: The Noise gate is enabled, with a short decay. This cleans up the sound with high-gain settings.
Distortion: The Teemah! distortion gives an edge when running the amp at lower Drive settings. You can also increase the intensity with high Drive settings by enabling Teemah!.
Amp+Cab: This uses the default settings, aside from dialing back the Drive. Doing so makes the distortion less intense, but gives Teemah! the space to kick the distortion into high gear. Parameters like Structure, Boost, Depth, Style, and Class tailor the sound further.
Post Amp+Cab EQ: Enable this to sweeten the sound by notching out some of the midrange around 4 kHz.
Double Take: This gives a more flanger, Eisley brothers type of sound. However, make sure you choose the correct source and channel mode in your DAW. For example, in Studio One I set the channel mode to stereo. This accommodates the stereo imaging in Double Take and the Dual Delay. However, the audio in the stereo channel is mono, so the Double Take's Source parameter is Mono.
Dual Delay: This provides the long delays used for leads, and opens up the stereo image.

EV Panama Blue (1.5)

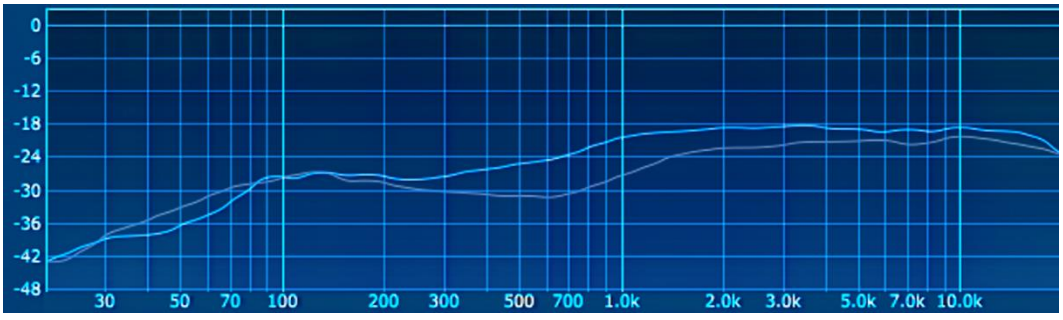
The Blue channel can hit sweet spots from clean to high gain, so it's ideal for crunch and overdrive—think punch rather than aggression. The Blue channel is more versatile than the Red channel.

Bass



This Bass control lives under 200 Hz, and doesn't dip into the lower mids. The main bump is at 100 Hz to 200 Hz, which brings up the guitar's lowest octave. Even at full cut, there's still a noticeable boost in this range.

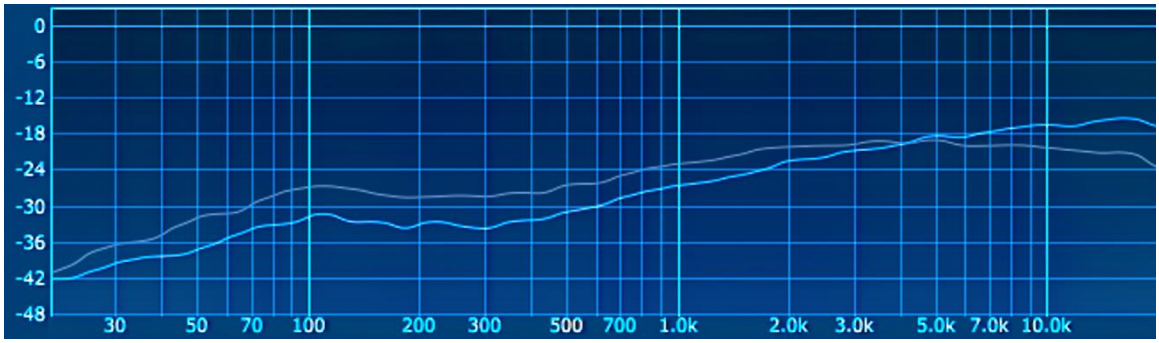
Mid



Mid picks up where Bass leaves off. Boosting gives the most emphasis in the upper midrange, but also gives a mild treble boost starting around 3 kHz. Cutting the mids scoops out in the range from around 150 Hz to 1.5

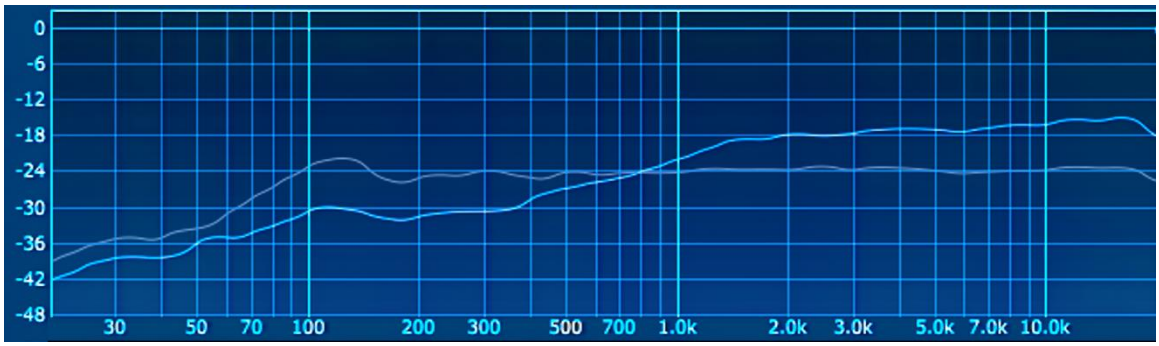
kHz. To emphasize the scoop, boost Bass to bring up the low frequencies and boost Treble to emphasize the highs. The Mid control has much influence on the sound. Turn up Mid to push the guitar more out front, or cut Mid to lay the guitar further back in an ensemble.

Presence



This is more like a bright switch that evolved into a control. Boosting increases treble from 5 kHz on up, while reducing midrange and bass below 5 kHz. Reducing lower frequencies gives a subjectively brighter sound. Cutting gives a flatter response.

High (Treble)



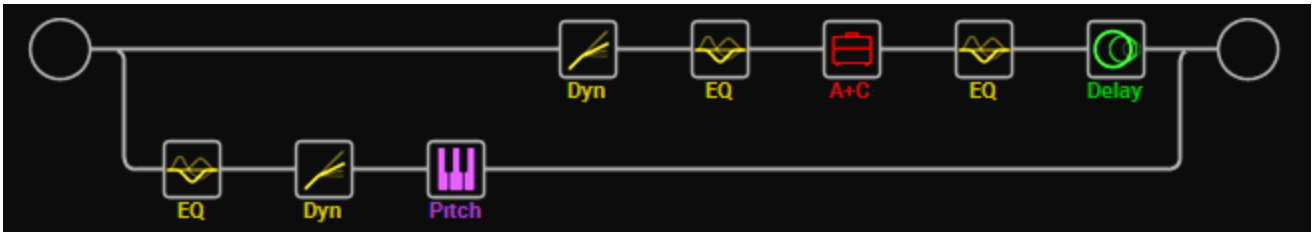
The High control has some “tilting” qualities. Turning up High almost acts like a shelving EQ above 1 kHz—the high frequencies get higher and stay higher. Meanwhile, frequencies below 1 kHz tilt downward. When cut, with other tone controls set to 5.0 the response is close to flat except for a slight bass bump around 125 Hz.

EV Panama Blue Preset

Barre Bet

If someone bets you can’t have bass play along with barre chords, take the bet and then load this preset. You may need to adapt your playing style a bit or edit some of the parameters, but to get started, play a full F-type barre chord cleanly. The root note will be doubled on bass. You can also play lead+bass lines over almost the lowest fretted octave (from open E up to the open D string).

This preset is level-sensitive. Hopefully it will work for you as is. If not, raise or lower the internal levels of some of the blocks that are close to the input. To lower the amount of bass, use the Boctaver’s -1 Oct slider.



The upper path is guitar, the lower path adds the bass. We'll deal with the bass path first.

EQ: The 10-Band Graphic EQ pushes the 62.5 and 125 Hz bands up full to +15, and sets the other bands and the output level to -15.
Dyn: This uses the 3-Band Comp to create a low band that isolates frequencies below 150 Hz. The band also compresses and boosts those frequencies. The other band levels are -60.0 dB. After passing through the 3-Band Comp, the highs have been cut so much that pretty much all that's left is a guitar's lowest octave.
Pitch: The Boctaver takes this mess o' bass and transposes it down an octave. This is how the preset generates a bass note along with the barre chord.

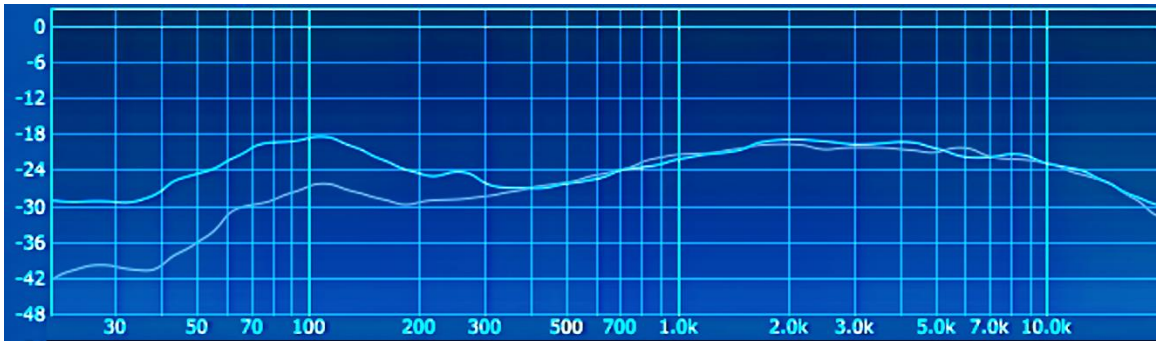
The guitar path opens with Dyn and Delay bypassed.

Dyn: To increase sustain, enable the Dyn block. It uses the Rochester Comp.
Pre-Amp+Cab and Post-Amp+Cab EQs: These do the same trick as described in the Creamy Lead preset for the German Xtra Blue. The first EQ cuts the response at 1 kHz, the post-Amp+Cab EQ increases treble to restore brightness but emphasizes sweeter high-frequency distortion components.
Amp+Cab: Compared to the default, this pulls back on the Gain, reduces High, decreases Sag for a sharper attack, and tweaks the Bias. The Cab is the default 4x12 XXL V30.
Delay: The Dual Delay emulates room reflections and creates a stereo image. Enable it for studio or Helix Native applications. You can bypass this block and replace it with something else for live use, because a live setup already has room reflections and you're probably not using a stereo setup.

EV Panama Red (1.5)

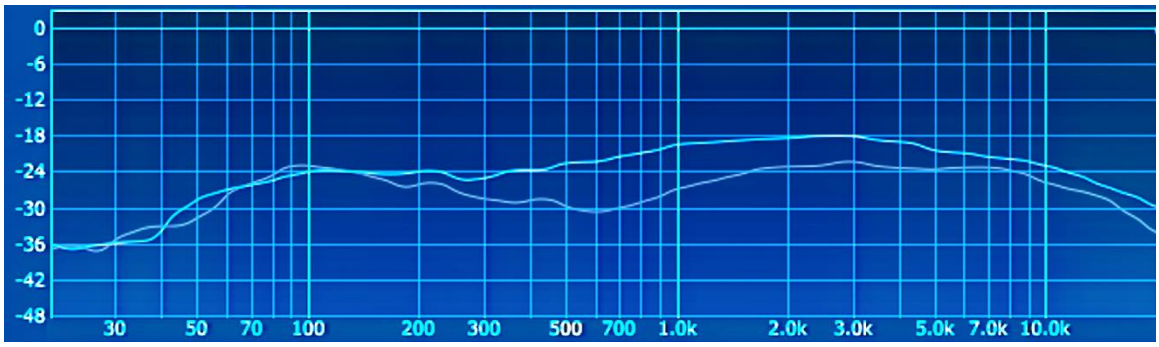
The Red channel zeroes in on metal/hard rock-friendly high-gain sounds, whether for huge power chords or sustaining leads. Crank up Gain to increase the sustain, or dial back Gain for more dynamic touch control.

Bass



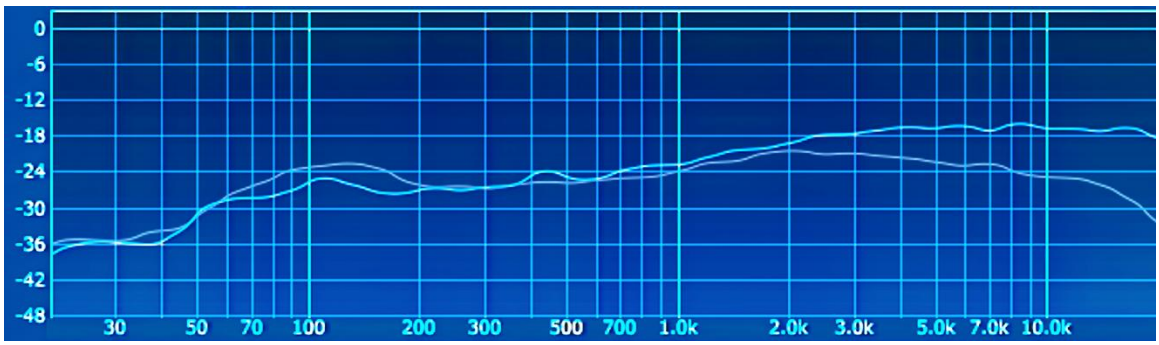
The Bass control is like a more severe version of the one in the Blue channel. The bass boost centered around 100 Hz is substantial. Even when cut, although the lower frequencies are lower starting below 300 Hz, there's still a bit of a bass bump. Think of the Bass control as the "make it thicker and heavier" control.

Mid



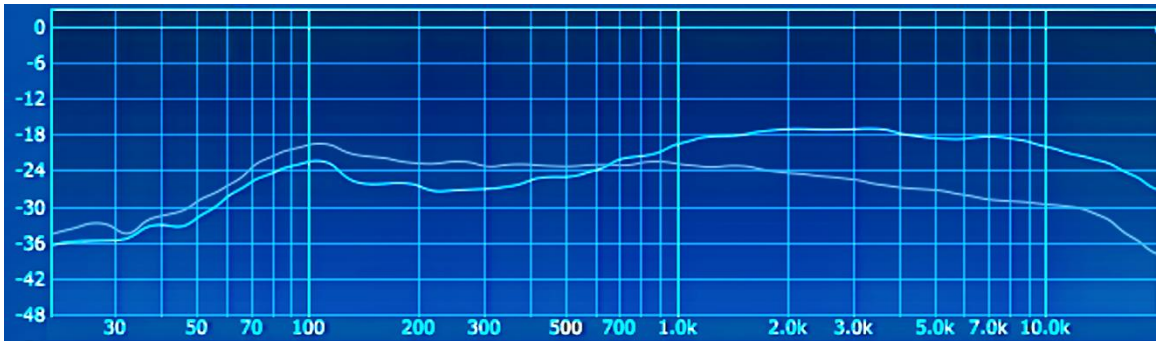
The curve is like the one in the Blue channel, with the following differences. Cutting scoops the mids a bit more. Boosting creates a high-frequency rolloff above 3 kHz, while cutting rolls off highs above 8 kHz. As with most mid controls, turn this up to push the guitar forward, or turn down to lay back.

Presence



This is more like a traditional Presence control than the one in the Blue channel. Boosting gives a treble lift above 2 kHz. Cutting starts a gentle high-frequency rolloff that starts around 2 kHz.

High (Treble)



Like the Blue channel, the High boost is almost shelf-like, with a fairly flat boost from 1 kHz to 10 kHz. The lower mids and bass also go down a bit to emphasize the highs. Cutting gently tapers off the high-frequency response starting around 1 kHz and retains a slight bass bump around 100 Hz.

For zinging treble, this is not the control you're looking for—Presence handles that job better. However, try setting Presence for the highs you want, then vary the High control to support Presence. Sometimes, turning down High while turning up Presence gives the best outcome.

EV Panama Red Preset

OTT Lead

OTT, as in “Over the Top”— distorted autowah, sustainer, dive-bomb whammy, echoes, chorusing, phaser, and plenty of gain. The preset opens with only the Amp and Cab enabled. Then, choose the effect(s) want to use.

Dynamics: The Rochester Comp provides additional sustain. For more conventional compression, the Red Squeeze compressor is a good choice. **Note:** the Rochester Comp pairs well with the Mutant Filter (see next).
Filt: The Mutant Filter is pre-distortion, so it sounds like you’re working a wah. The Rochester Comp before it smooths the response to your playing. **Important:** Use the Rochester Comp’s Gain parameter to obtain the Mutant Filter sweep range that best matches your dynamics.
Amp: Compared to the default, this lowers Sag for more attack, cuts Ripple, increases Presence, reduces High, and edits Bias for maximum gain.
Cab: The Dual Cab combines the 4x12 Cali V30 with the 4x12 SoloLeadEM. Both use a 906 Dynamic mic.
Pitch: The Whammy block is Mono, so it’s before the stereo Dual Cab. Using Mono saves enough processing power to add the other effects. The Whammy’s Heel pitch is 0 and the Toe pitch is -24 semitones. It seemed more logical to push down on the pedal when you want the effect, but you may prefer setting the Toe to 0 and Heel to -24 semitones so pulling back on the pedal creates the “dive bomb” effects.
Mod: The Deluxe Phaser is mono to conserve processing power.
Delay: The Dual Delay has over-the-top settings to match the preset’s theme. The delay times are long with the delays mixed high, there’s a fair amount of feedback, and the Chorus modulation mode is enabled.
Verb: Stereo Dynamic Ambience adds an acoustic space to the sound.

Chapter 3 | PRESET CONSTRUCTION TIPS

Creating your own presets is a lot of fun, but it can also help to see how others approach creating custom presets. These presets (with free presets in the Free Files folder) show some novel Helix techniques.

Talk Box

The **Talk Box** preset isn't an *exact* emulation of a talk box—but it gives that vibe. An Expression Pedal varies the frequency of three Parametric EQ stages. These are set to the highest possible Q and Gain (fig. 3.1).

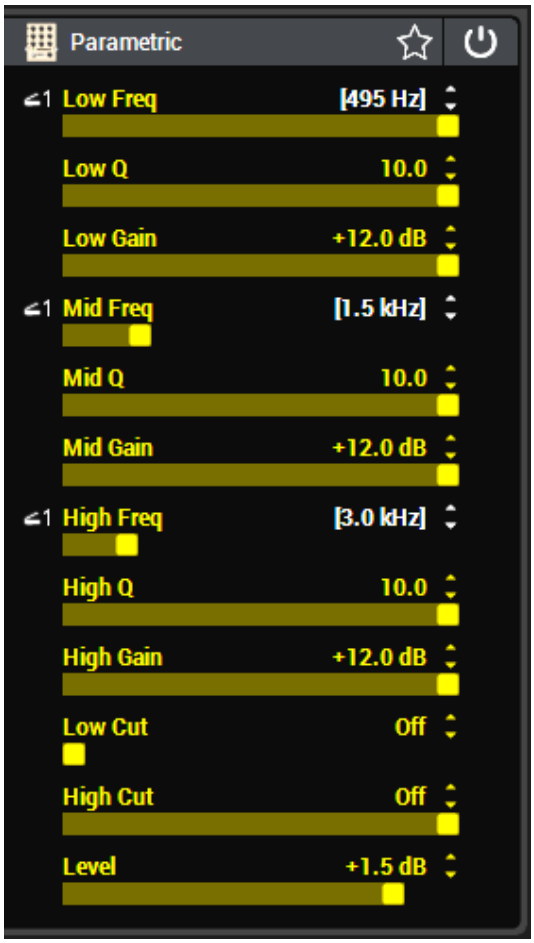


Figure 3.1 Talk Box parameter settings for the Parametric EQ block.

Fig. 3.2 outlines the preset in orange. The Gain block creates a parallel path and enables proper bypassing. In the Mixer, invert Path B's Polarity. The Path A and B levels must be equal and panned to center.

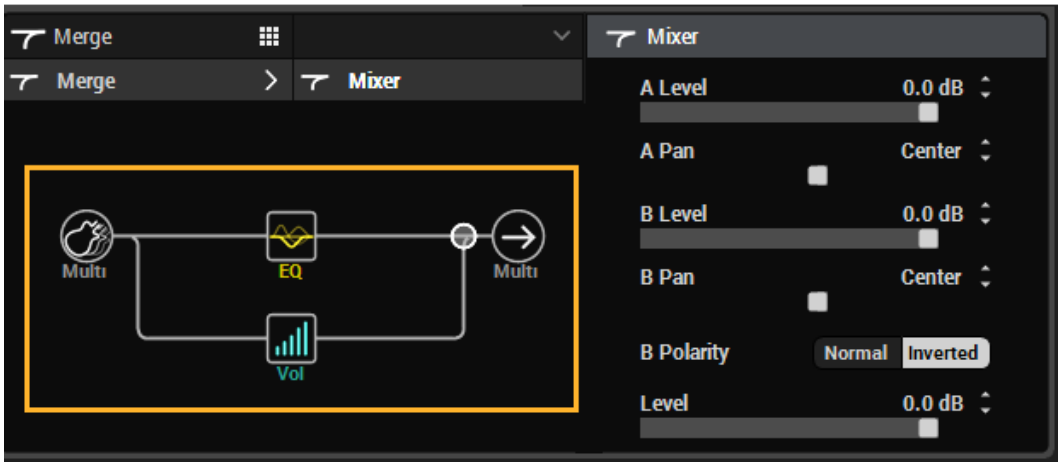


Figure 3.2 Effect routing and Mixer settings.

This preset controls three parameters with a pedal. Because the preset is EQ-based, and adding a Wah block automatically assigns its Position parameter to EXP 1, you'll probably want to use EXP 1 for this preset.

The pedal (fig. 3.3) varies the Low Freq from 250 Hz to 495 Hz, the Mid Freq from 750 Hz to 1.5 kHz, and the High Freq from 1.5 kHz to 3.0 kHz.

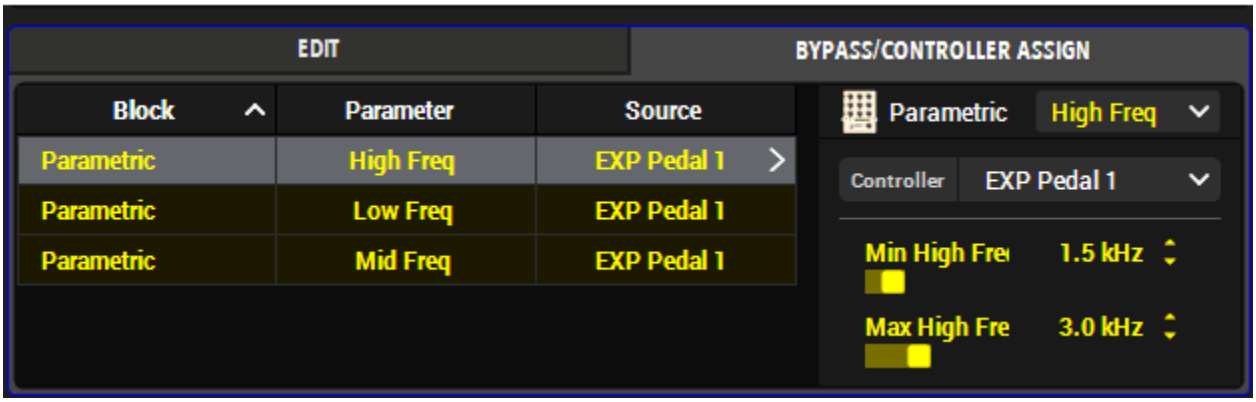


Figure 3.3 Assign the Parametric Freq parameters to EXP Pedal 1.

To assign parameters to the pedal:

1. Press the Home button and select the EQ Parametric block.
2. Press and hold the Low Freq parameter knob until the Controller Assign screen appears.
3. Push on the Learn Controller knob and move the EXP 1 pedal.
4. Press the Home button again. Repeat steps 2 and 3, but this time, press and hold the Mid Freq parameter knob instead of the Low Freq parameter knob.
5. Repeat step 4, but this time, press *Page* > once. Then, press and hold the High Freq parameter knob instead of the Mid Freq parameter knob.

Bypassing the Talk Box

If you bypass the Gain block and EQ at the same time, the audio will cancel because the two paths will be out of phase. This is because Mixer's B Polarity is inverted. So, set the Gain block to -120 dB. Program the footswitch for this effect to enable the Gain block when *bypassing* the EQ block. This turns the parallel path down by -120 dB, so the audio no longer cancels, and you hear the dry guitar.

Enabling the EQ bypasses the Gain block, which restores the parallel paths to give the Talk Box effect.

Parallel Autofilter

Autofilters are cool, but they have two main issues:

- They thin out the sound. To avoid this, use an Autofilter as a parallel effect by setting the Mix control to less than 100%.
- The optimum settings depend on many variables—the input level, your playing style, the intended result, and the effect parameter settings.

Because of the variables, the **ParallelAutoFilt** preset is intended as a starting point. Fig. 3.4 shows Path A's filter settings on the left, and Path B's filter settings on the right. Edit these settings, as described under Customization, to accommodate the dynamics of your playing.

The Mixer pans the two paths oppositely for a stereo effect. The preset collapses perfectly to mono anyway.

Figure 3.4 Filter settings for the two Autofilter paths.

Customization

- Set the Attack to at least 99 ms. This avoids a potentially “warbly” sound. It’s caused by the Attack being too fast to smooth out your dynamics.
- The Attack and Decay times for the two paths are different. This gives a looser feel to the tracking, which creates a more varied sound.
- To make the Autofilter Sensitivity settings less critical, the dynamics processor at the preset’s beginning smooths out the dynamics somewhat.
- The crucial parameters for tracking dynamics are Sensitivity and Frequency Depth. You’ll need to edit these, because the preset is almost certainly not set up for your input levels, how hard you pick your strings, your string gauge, or playing style. Once you nail these parameters, save the preset.
- To accentuate the stereo image, change both Autofilters’ Mode to High Pass.
- Changing Path B’s Mode to Low Pass allows more low frequencies to pass through, which gives a bassier sound.
- For up and down parallel sweeps, try these Path B settings: Direction = Down, Mode = High Pass, Frequency = 100 Hz, Freq Depth = 5 kHz.

Through-Zero Auto Flanger

Sometimes a block can provide an unexpected function. The **Thru0 AutoFlange** preset (fig. 3.5) transforms the Pitch Wham into an automatic, through-zero flanger.

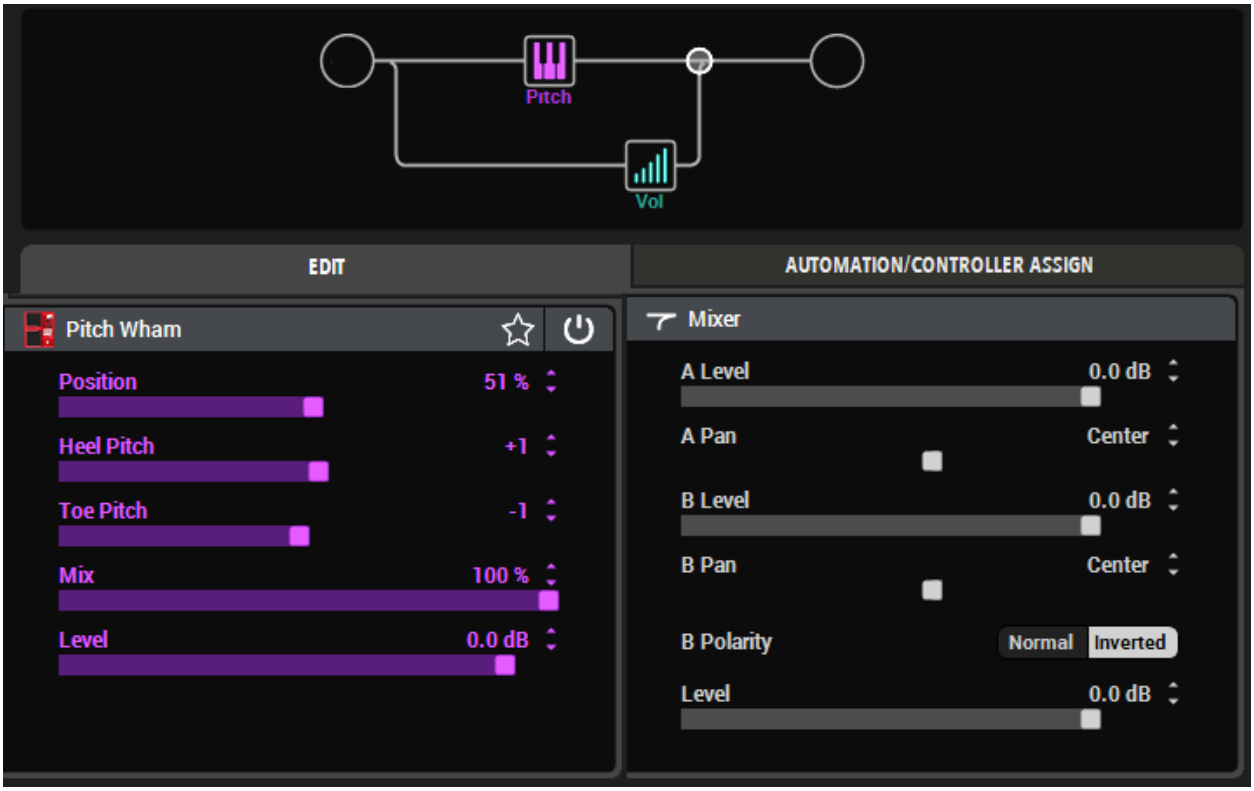


Figure 3.5 The Pitch Wham is not being used as a Pitch Wham.

Tech Talk: Through-Zero Flanging

Flanging was originally done with two tape recorders that played back the same audio, with their outputs mixed together. Changing the speed on one recorder caused the music to creep ahead, or lag behind, the music playing back from the other recorder. If the two recorder outputs were out-of-phase, and one recorder played back the same audio at the same time as the other recorder, the audio would cancel briefly. This was called the “through-zero” point.

How It Works

Flanging occurs because the Pitch Wham’s pitch inherently varies over a small range. You don’t need to move the pedal for this to happen. The Vol (Gain) block exists solely to create parallel Path B. To obtain through-zero flanging, the Mixer inverts Path B’s Polarity, so the audio cancels when the two paths play back the same audio at the same time.

Setting A Pan and B Pan to center produces the most intense through-zero effect. Offsetting them somewhat (e.g., Left 40 and Right 40) gives psycho-acoustic panning effects in stereo, as well as flanging. A mono output produces flanging only, regardless of the pan settings.

To bypass the effect, set the Volume block to -120 dB, and enable/disable the Volume and Pitch Wham blocks *oppositely*. See Bypassing the Talk Box for information on why this is necessary.

Use Preamps to Save Processing Power

Line 6 recommends using the Preamp modules to feed external amps. That's an obvious application, but to my ears, some Preamps have a more focused distortion sound compared to the standard amps. In fact, when feeding a cab, and tweaking the Preamp EQ slightly, I sometimes *prefer* the Preamp+Cab combination.

Fig. 3.6 shows a practical example. In both cases, the Cab is a 4x12 Greenback 25. The preamp, and amp in the Amp+Cab block, are both Brit Plexi Jumps. This preset retains the essential amp sound, but with enough DSP power left over for plenty of effects, even with a power-hungry amp like the Brit Plexi Jump.

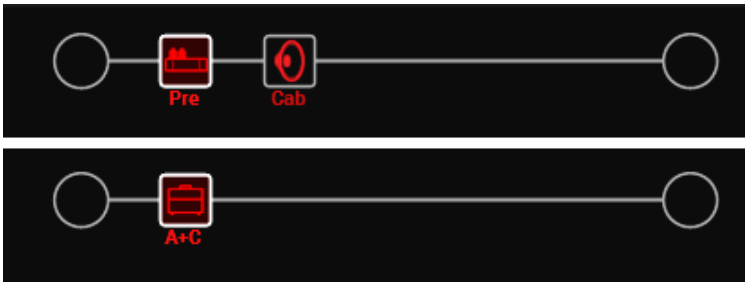


Figure 3.6 These presets sound almost identical, when adjusted for the same perceived volume.

In fig. 3.7, the Poly Capo block provides transposition for low tuning effects. The top preamp and bottom amp are both Brit Trem Brt, and both use a 4x12 Greenback 25 Cab. To demonstrate the difference in DSP power consumption, the upper path lets you choose from all Legacy and mono distortions, or 34 stereo distortions. The lower path requires more processing power. You can choose from all Legacy distortions, but only 33 mono or 13 stereo distortions.

Figure 3.7 The two paths sound virtually identical (although I slightly prefer the top path distortion sound).

Preamp Downsides

The preamps have inconsistent output levels, because Line 6 models the preamp output before it goes to the power amp being modeled. So, the preamp levels reflect differences in the preamps themselves. This makes sense, given that their intended application is to drive external amps. To drive whatever amp you're using, Line 6 leaves it up to you to add Gain blocks that provide enough gain.

This application usually requires bringing up the preamp levels so that they're useable in Helix as pseudo-amps. Turning up **Ch Vol** can often equal typical amp levels. If not, then also turn up the cab's **Level**. Or, if you follow the cab with an effect that's always enabled (like EQ), raising its **Level** parameter value might be enough. If all else fails, add a Gain block after the preamp.

With really low-level preamps, you may need more gain than a single Volume/Gain block can provide. I rarely use the following Preamps (listed from higher to lower output) with HX Stomp, because compensating for their low levels requires two extra blocks. Using two blocks may be too much of a tradeoff to save DSP power. But if preamps give the desired sound, I don't mind the extra blocks—they require very little DSP power. Besides, with a 4-path Helix, Paths 1 and 2 can go in series to increase the potential number of blocks.

- Placater Dirty
- ANGL Meteor
- Essex A30
- German Ubersonic
- Jazz Rivet 120
- Archetype Lead
- Divided Duo
- PV Panama
- EV Panama (Blue and Red)
- Archetype Clean
- Essex A15
- Line 6 2204 Mod
- Line 6 Badonk
- Mandarin Rocker
- Solo Lead Clean
- Cali Rectifire
- Matchstick Ch1
- Solo Lead (Crunch and OD)
- Line 6 Fatality
- Line 6 Epic
- German Xtra Red

Tip: With a parallel effects chain, the Mixer's Level parameter may be able to add the necessary amount of gain for a preset, so you won't need a Volume/Gain block. Or, it can serve as a second Volume/Gain element that doesn't require adding another block.

Bi-Amped Brit Plexi Jump Preset

The **BiAmpBritPlexJmp** preset (fig. 3.8) gives a preview of the bi-amping techniques in Chapter 5.

This preset also shows how much processing power Preamp blocks can save, compared to using amps. The Brit Plexi Jump Amp+Cab is one of Helix's most processing-hungry blocks. To duplicate the bi-amped preset below with Brit Plexi Jump Amp+Cabs in parallel, you couldn't use the effects below except for the stereo Parametric.

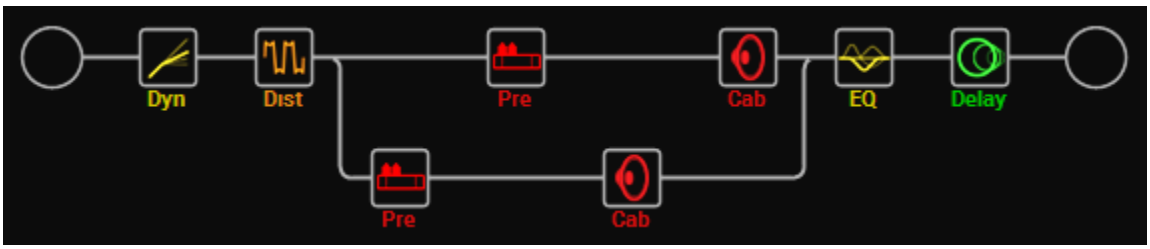


Figure 3.8 Use preamps for the sound of two Brit Plexi Jump Amps in parallel *and* multiple DSP-hungry effects (LA Studio Comp, KWB Distortion, stereo Parametric EQ, stereo Adriatic Delay).

Clean Sounds with “Dirty” Amps

For a different kind of clean sound, start with a dirty one, and pull *way* back on the input level or Drive control. Although this approach doesn’t work too well with a *really* dirty amp like the German Mahadeva, the following Amp+Cab models are good choices for this application:

- Soup Pro
- Both Tweed models
- All US models
- Both Essex models
- Most Brit models
- Cali models (except for Lead and Rectifire)
- Solo Lead Clean and Crunch

Here’s the technique:

1. Turn down the **Drive** control (outlined in white in fig. 3.9) below 3, and sometimes even below 1.
2. To compensate for the volume drop, increase the **Ch Vol** parameter (outlined in orange). This control does not alter the tone.
3. Adjust the **Master** parameter (outlined in blue) to change the level. This also alters tone, so between Drive, Ch Vol, and Master, you should be able to obtain the clean sound you want. For a hotter output level, turn up the Cab Level, or insert a post-amp Volume/Pan block, set it to the Gain function, and increase its level.

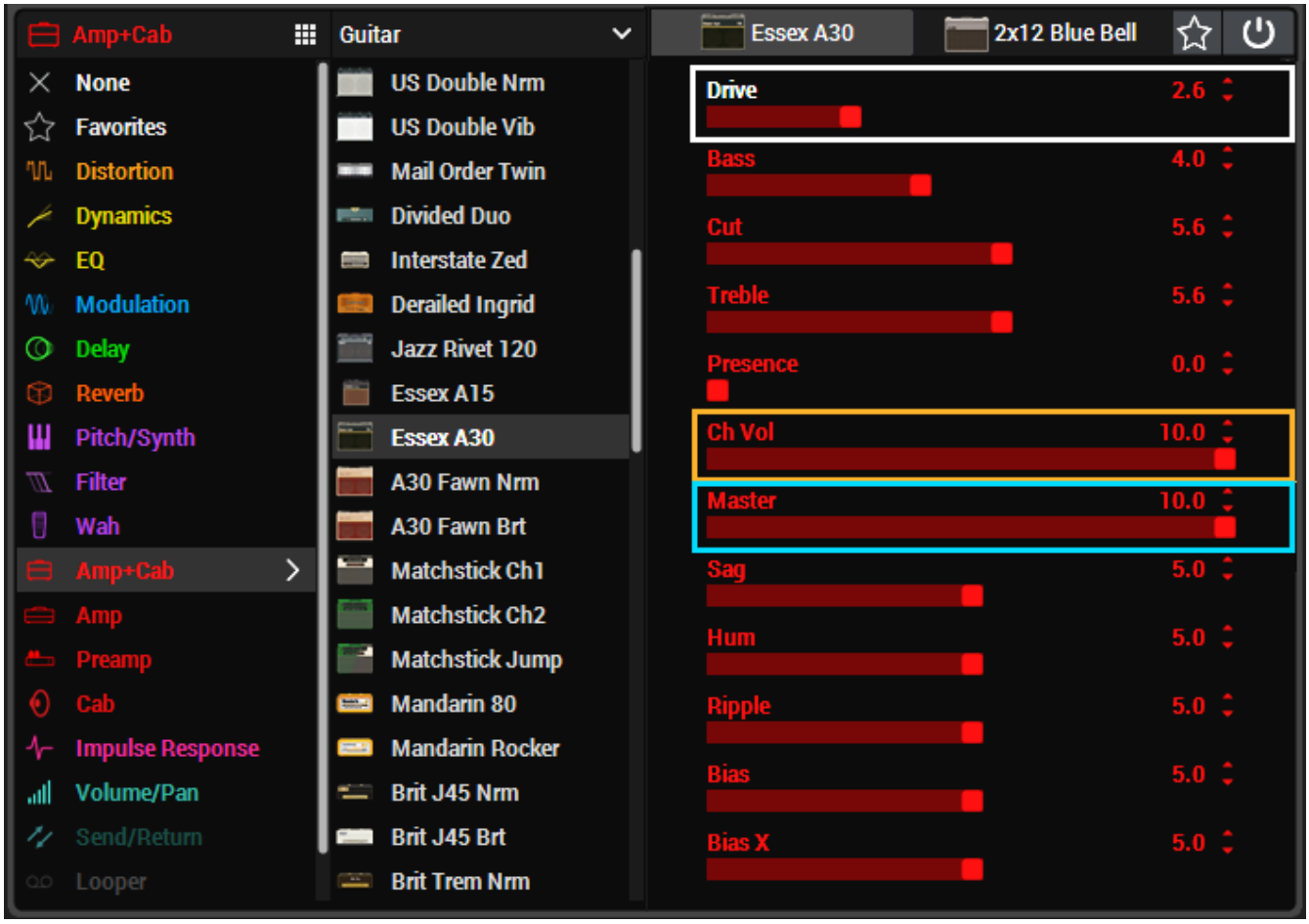


Figure 3.9 These three parameters can coax clean tones out of otherwise “dirty” amps.

50s Vintage Echo Amp

The Helix offers several ways to recreate the sound of slapback echo and ancient amps. This **Vintage Echo Amp** preset is an "effects mashup" that shows another side of Helix. I was inspired by the Echosonic amp (which had a built-in tape echo unit), tube preamp experiments I did back in the day, and the famous Sun Studios echo sound.

Vintage amps with built-in effects (like tremolo) typically wired the effects between the preamp and power amp. So, the preamp feeds a Retro Reel for the tape sound, followed by the Simple Delay set to 135 ms for the slapback delay (fig. 3.10). There's no power amp because the preamp provides the vintage character. It's subtle, but responsive. The cab completes the amp feel, and the Dynamic Ambience reverb provides a little ambiance.

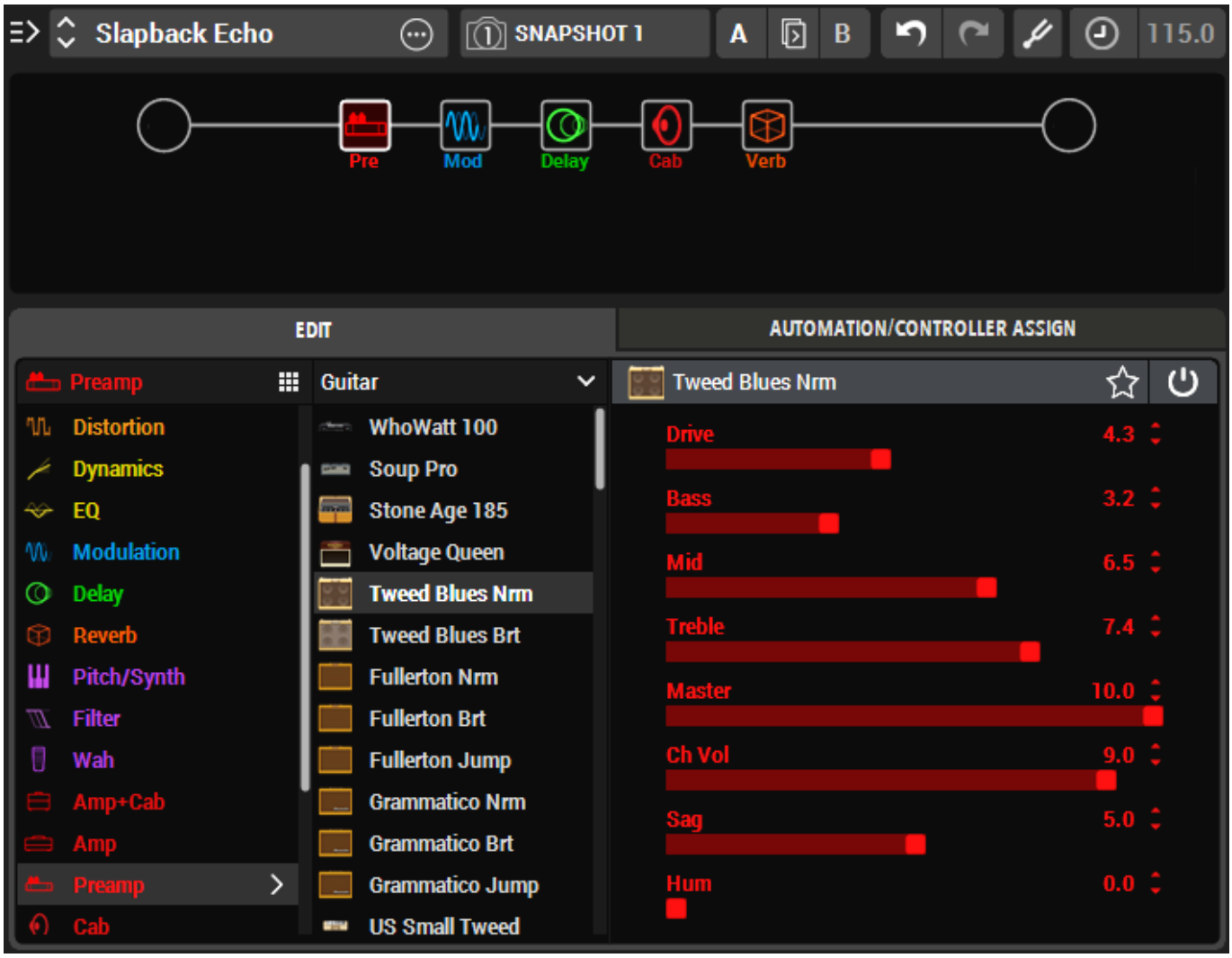


Figure 3.10 Using the Preamp for the main amp element gives this preset a unique character.

To make sure this wasn't re-inventing the wheel, I tried other, various conventional approaches (echo after an amp, before an amp, using the Transistor Tape delay, and so on). There was just something about this configuration that I liked.

I'm not saying it's some scientific breakthrough—but think "1950s rockabilly," load up this preset, and travel back in time.

“Blooming” Reverb

A reverb’s traditional predelay parameter keeps reverb from “stepping on” note attacks, but it can add an echo that sounds overly percussive. With the preset in fig. 3.11, the reverb “blooms” after any initial note attacks, which creates a lush, gorgeous wash of reverb. The sound is smoother than using predelay.

The **Blooming Reverb** preset is based on my Hot Springs reverb (no relation to the one in Helix), a DIY project from the Mesozoic era of recording. Its goal was to reduce the “sproinnng” when the spring reverb kicked in.

Figure 3.11 Inverting a mixer channel is the key to making reverb bloom. Crucial parameters are outlined in white.

How It Works

This technique inserts reverbs with almost identical settings in parallel. The Mixer inverts Path B’s Polarity, so if the reverbs have the same settings, the sound cancels. However, suppose you increase the decay time for one

reverb, but not the other. Although the initial reverb sound cancels quite a bit, the reverb blooms and expands as the reverb tails become increasingly dissimilar.

However, with a 2-Path Helix, the dry sound cancels too (see below for setup with a 4-Path Helix). So, initially set both Mix parameters to 50%. For more dry sound, increase the value for one of the Mix parameters. For some reason, this doesn't have too much impact on the way the reverb cancels.

The Legacy reverbs, which generally require the least processing power of any reverbs, work well with this preset. The ones with longer decay times (Plate, Chamber, Hall, and Cave) work best. When processing power is no object, the stereo Double Tank is a good choice.

Because of the initial cancellation, the overall reverb level may be relatively low. The post-mixer Gain block increases the level as needed.

Bypassing Issues

With HX Stomp, bypassing the reverb effect is complicated. For a dry sound, you need to bypass the two reverbs, the gain block, and change the Mixer's Path B Polarity from Inverted to Normal. So, I recommend using this technique in a stand-alone preset, with any other needed blocks before the Split, and/or after the Mixer. For example, if you want a sweet, clean guitar sound, insert a compressor block and a clean amp before the Split, and maybe EQ after the mixer to shape the tone further.

With a 4-path Helix, implement this technique like a traditional send effect. As with HX Stomp, the reverbs go in one path. The other path provides the guitar sound. However, with this routing, set both reverb Mix controls to 100% reverb. Edit the Gain at the end of the reverb path to alter the blend of reverb with guitar.

Dynamic Brightener

Humbucking pickup design minimizes hum, but the tradeoff is a kind of dullness that, although it works well for distortion, has less presence with dry, clean guitar sounds. Furthermore, compared to acoustic guitars, all electric guitars (regardless of the pickup type) have less perceived dynamics. This is because the tone doesn't become brighter when you play the strings harder.

Fortunately, it's possible to impart more presence and sparkle to electric guitars, so that their dynamics are more like an acoustic guitar. The key is the **DynamicBrighten** preset's parallel path (fig. 3.12):

- Path A is the dry guitar.
- Path B uses the Noise Gate to let through only the peaks of your playing. Those peaks then pass through an EQ, set for a bass rolloff and a high-frequency boost, so that the peaks have a brighter sound. When mixed with the dry guitar, loud peaks sound brighter.

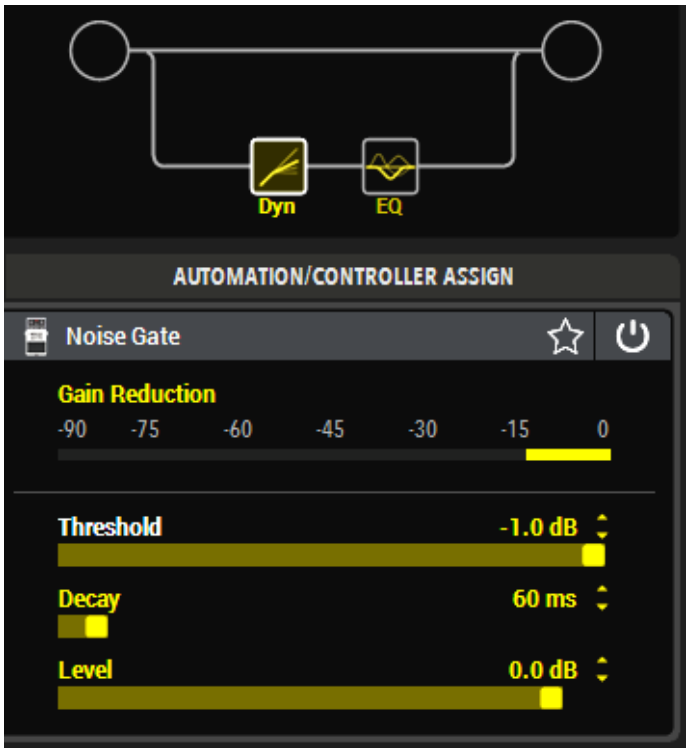


Figure 3.12 The parallel path emphasizes peaks by boosting the highs.

This preset's effectiveness depends on your guitar, strings, and playing style. Here's how to set it up:

1. In the Mixer, temporarily turn down Path A's level. This makes it easier to adjust the block parameters.
2. Bypass the EQ. Edit the Noise Gate settings so that it passes only the peaks of your playing. The optimum **Decay** time depends on your playing style's percussiveness. If the sound is too choppy, lengthen the Decay.
3. You probably won't get the right sound immediately—the settings are tricky. The Noise Gate's **Threshold** is the crucial parameter. After it grabs only the peaks, then experiment further with the Decay setting.
4. The EQ has a lot of high-frequency gain. If there's distortion, lower the 4, 8, and 16 kHz levels (try +10 or +12 instead of +15). Also, adjust the EQ's level for a subtle brightness. Too many highs sound screechy.

Bypassing both blocks is usually sufficient for returning to a dry sound.

Taming the Tycoctavia

The good news: Helix does a good emulation of the Tycobrahe Octavia. The bad news: Helix does a good emulation of the Tycobrahe Octavia. This distortion was optimized for single-note leads played high on the neck. It generated an octave-higher harmonic—along with a lot of fizzy artifacts. It helps to use the neck pickup. But we can help it even further, as you'll hear when you load the **Tyco Tamer** preset.

The key is reducing the highs going into the Tycoctavia. Then, it can concentrate on the note's fundamental, instead of being distracted by harmonics and "string zing." Trimming the lows helps even more, by making it possible to play notes lower on the neck with acceptable tone.

The post-Tycoctavia simple EQ isn't strictly necessary, but adding a little brightness can compensate for rolling off the highs before going into the Tycoctavia. Fig. 3.13 shows the parameter settings.

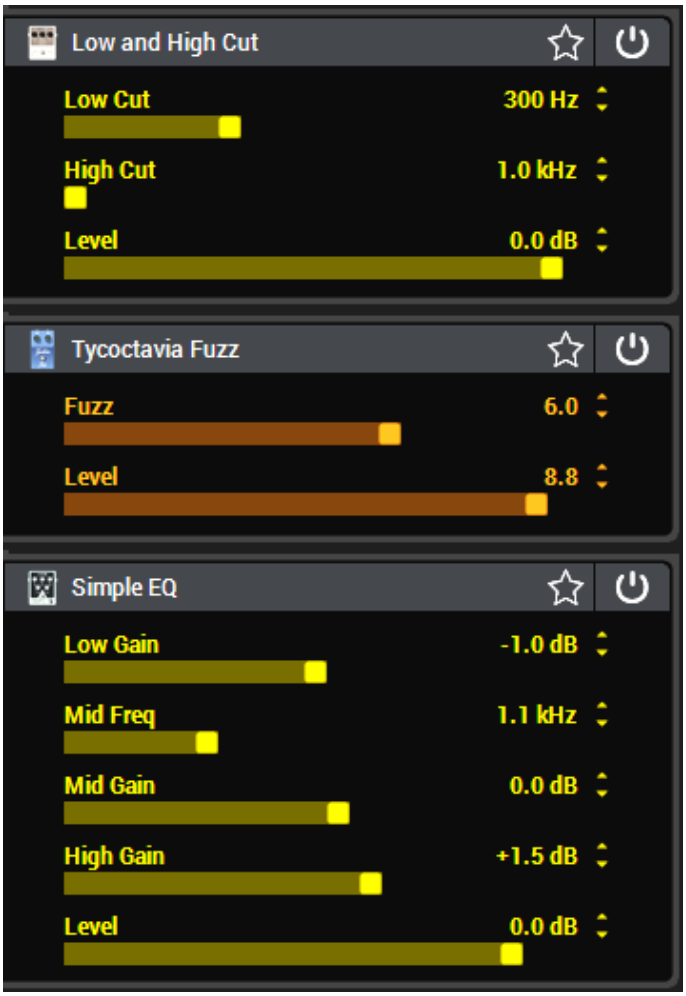


Figure 3.13 The **Tyco Tamer**'s parameters, with signal flow from top to bottom.

For another way to obtain octave-higher distortion, check out the preset 3B.+OctaveUp.

The Secret Wah

Personal bias alert: I'm not a big fan of wah pedals, because their stereotyped sound takes over a track. I was hoping Helix would have a kinder, gentler wah sound—and it does, if you know where to look.

When programmed appropriately, the Simple EQ makes a unique wah sound. I must admit, this wasn't something smart that I figured out, but a happy accident. I was trying to find effects that can produce saturation when not being used as effects, and in the process, stumbled on this sound.

Set all the Gain controls to maximum (fig. 3.14) and assign the Mid Freq parameter to an Expression Pedal. To maintain unity gain through the Simple EQ, reduce the Level to compensate for the added gain. The sound is more liquid and less gimmicky than a standard wah.

Figure 3.14 The Simple EQ creates a different type of wah sound.

Tech Talk: Why It Sounds Different

A conventional wah has a peak, and the low and frequencies roll off on either side of the peak. The Secret Wah has a dip in the response at 500 Hz, because the Low and High Gain controls boost the highs and lows. Varying the Mid Freq produces a mild peak that slides above and below this response dip. So, you still hear the filtered wah sound, but you don't lose the low and high frequencies. Furthermore, the dip allows the wah's peak to sound more prominent than you might expect.

The Customizable DIY Wah

After seeing how to create a different kind of wah sound in the previous tip, let's explore a wah with a customizable filter peak, customizable frequency range, and an authentic wah sound. The **Custom DIY Wah** preset came about because I wanted a wah with a very broad peak to insert before an amp. None of the Helix Wahs did quite what I wanted—so it was time to warm up the virtual soldering iron (fig. 3.15).

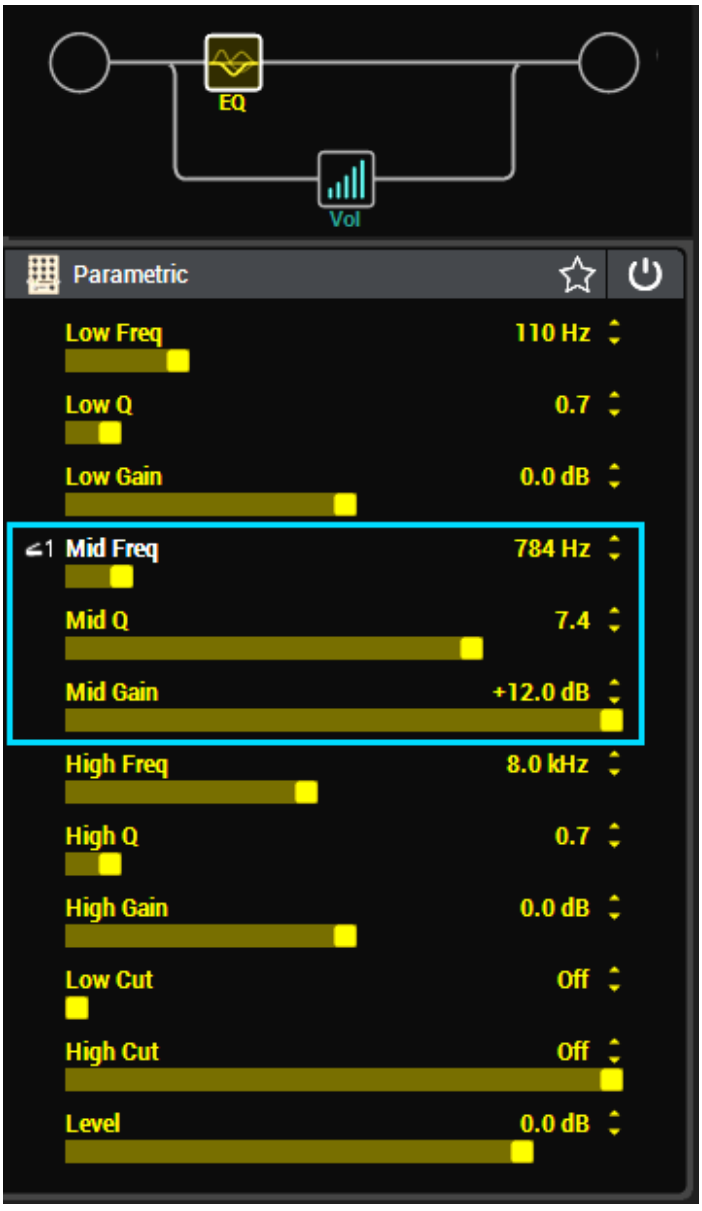


Figure 3.15 How to create a custom Wah. The only parametric EQ parameters that matter are outlined in blue. Leave the rest at their defaults.

As mentioned in the previous Tech Talk, a conventional wah has a peak, and the low and high frequencies roll off on either side of the peak. The Parametric EQ doesn't have this kind of response, so we'll need to create it.

Route the Parametric EQ in parallel with a Volume (Gain) block. This block creates a parallel path, because this effect needs an output Mixer that can invert polarity.

At the Mixer, by setting Path B's Polarity to Invert (fig. 3.16), audio going through the Parametric that *isn't* being filtered by the Mid EQ cancels out. So, everything except the wah peak cancels, which gives the traditional wah frequency response characteristics.

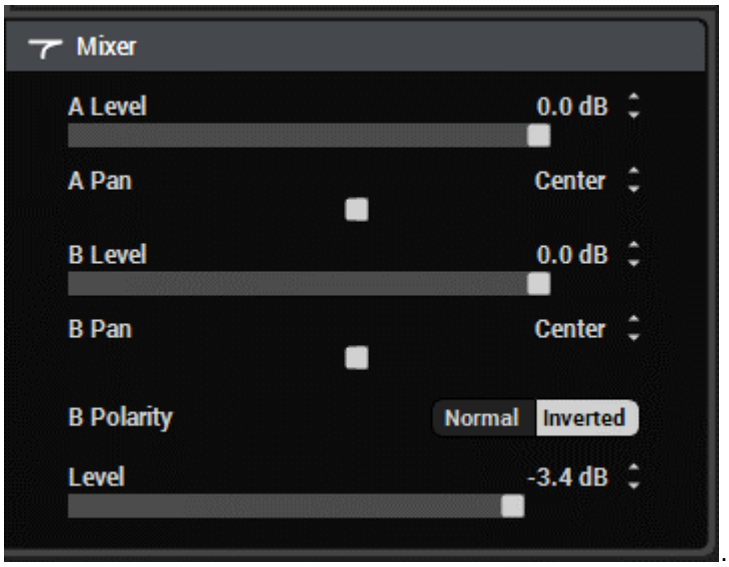


Figure 3.16 Output Mixer settings for the customizable wah.

How to Use the Customizable Wah

1. Assign an Expression Pedal (EXP 1 is the default for Wah) to the Parametric EQ's Mid Freq parameter.
2. Set the Mid Freq's high- and low-frequency limits as desired, using the Bypass/Controller assign panel.
3. Set Mid Q for a suitable amount of resonance.
4. Set the Mid Gain close to Max. This compensates for the lower level due to thinning the sound with the filter.

That's all there is to it. However, not having the two paths *exactly* out of phase diminishes the wah effect. To test this, set the Parametric EQ midrange Gain to 0.0. You shouldn't hear any sound coming through the Customizable Wah. If so, check for a level mismatch from:

- The Input block sending different amounts of signal into the two parallel paths.
- The Mixer levels not being matched.
- The Volume block and Parametric output Levels not being set to 0.0.

This effect excels for a parked (cocked) wah sound when it's inserted before an amp with lots of distortion. Set the Mid Q to around 0.5 to 1.0, and you'll be able to move the pedal, give a wah effect, but not have the breaking-up sound of a traditional wah (unless you kick the Q way up).

Bypassing

You can't bypass the Gain block and EQ at the same time. The two paths will be out of phase, so the audio will cancel. The solution is the same as with the Bypassing the Talk Box: set the Vol Gain block -120 dB, and assign the footswitch so that bypassing the EQ block enables the Gain block. This turns the parallel path down by -120 dB, so the audio no longer cancels, and you hear the dry guitar. Enabling the EQ bypasses the Gain block, so this restores the parallel paths.

Dual Tremolo

Tremolo was big in the 50s and 60s, especially in surf music—so it has a stereotyped sound. But why be normal? The Tremolo/Autopan goes beyond what antique tremolos did, and the **Dual Tremolo** preset (fig. 3.17) takes advantage of this by inserting two Tremolos in series.

Figure 3.17 Following a faster tremolo by a slower one, and setting them to Note Sync, creates complex rhythmic effects.

In this context, Note Sync is important, because you want the two tremolos to work together rhythmically.

Tip: Remember that if they don't start at the same time or match up quite right, a quick push on the Tap footswitch, or sending a MIDI controller #64 message, will sync them.

The first Tremolo runs at a fast rate, like 1/16th notes. Square wave modulation works well for a "chopped" sound, while a sine wave gives a smoother, more pulsing effect. The second Tremolo runs at a slower rate. For example, if it syncs to half-notes, the second tremolo lets through a string of 1/16th-note pulses for a half-note, then attenuates the pulses for the next half-note.

In the preset, the second Tremolo syncs to eighth notes. The Saw Down waveform imparts a percussive effect to the 1/16th-note pulses from the first tremolo. This is just one of many options. Try different rhythms, different waveforms, and different duty cycles.

E Drone Raga

I'm fascinated by Indian classical music, so this preset (fig. 3.18), as well as the next one, are homages to the genre. You play your guitar through Path A, while Path B generates a synthesized, tamboura-like drone. It's the kind of preset where after a long day, you can put on your headphones, play some evocative, beautiful lead lines, and bliss out. The Distortion block in the guitar path is optional. This preset also *loves* E-Bows.

Figure 3.18 The upper path handles your guitar, while the lower one generates a synthesized drone.

The Drone

The 3-Note Generator Pitch Block is the heart of the drone section. The three sawtooth wave notes are set to E (octave 1), B (octave 2), and E (octave 3). The tone has a metallic, tamboura-like sound.

The Spin Cycle (Legacy modulation) adds slow, cyclic filtering, while the Tremolo adds slightly faster amplitude changes. The different speeds keep the sound hypnotic rather than static. Finally, the Legacy Particle Verb effect creates a distant, ethereal feel.

The Guitar Path

The Deluxe Comp provides sustain for long, languid lead lines with lots of bending. Try the neck pickup, and perhaps pull back the tone control.

The Compulsive Drive has a relatively sweet fuzz sound, but an unprocessed guitar can give a peaceful, pure feeling. The preset uses the Pillars distortion that was introduced in version 3.50.

The Delay provides stereo spaciousness. The Dynamic Hall at the end gives a classy reverb sound that makes the guitar even more evocative.

Customization

There’s a lot you can do with this preset:

- Increase the Delay time and Feedback parameters for a looper-like vibe.
- Try different distortion options. I was going for a relatively clean sound, but some might enjoy pairing a nasty distortion sound with the drone.
- Vary the Particle Verb’s Dwell parameter. The higher the setting, the more abstract the drone.
- 50% Mix for the Particle Drone seems about optimum. Raising the Dwell and Mix parameter values gives a more transcendent feel.

Hindustani

Hindustani music is the classical music associated with the northern part of India. It's melody-based, with no harmony as the term is understood in western music. It also leans heavily on improvisation. The **Hindustani** preset requires a 4-Path Helix device (fig. 3.19 shows this set up in Helix Native), which allows taking the concept behind the E Drone Raga to another level.

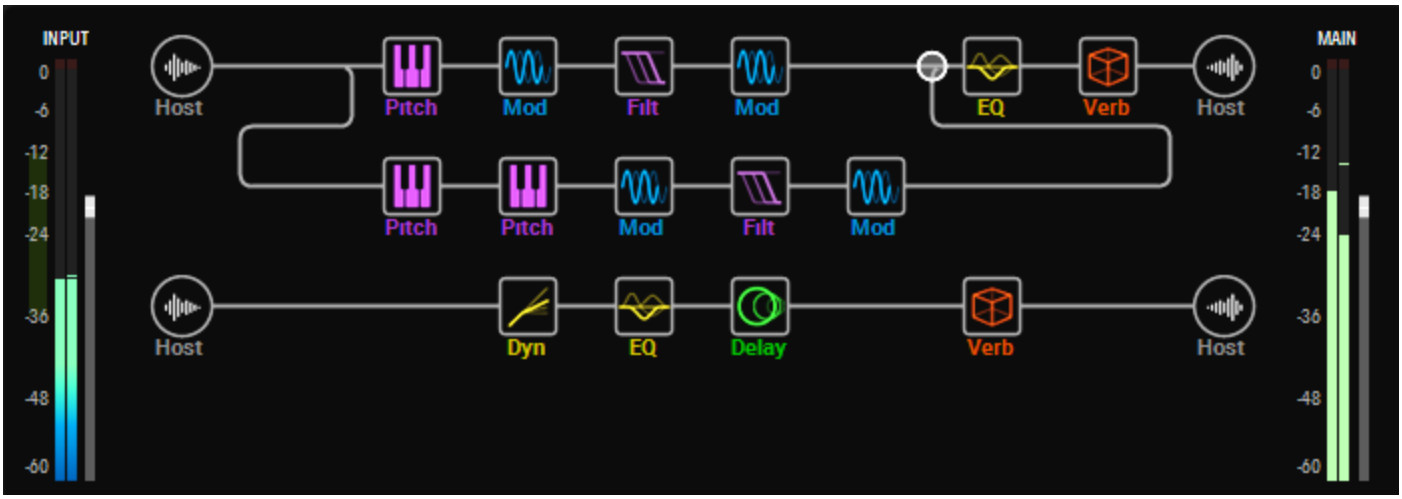


Figure 3.19 Path 1 emulates an electronic tamboura, while the guitar plays through the processors in Path 2.

Path 1 is dedicated to sound generation. It uses a pair of 3 Note Generators. Your guitar plays through Path 2.

Path 1 emulates the tambura-based drone that's common in Indian music. Each oscillator goes through a Tremolo, followed by an Autofilter. The filter frequency reacts to amplitude changes from the Tremolos. The Tremolos after the Autofilters add slow, asynchronous level changes to augment the hypnotic effect. Paths 1A and 1B are panned oppositely in the Mixer.

The second Pitch block in Path 1B is a Simple Pitch. It detunes the pitch by -10 cents and adds 60 ms of delay. If you wear headphones, the detuning and delay make the sound swim around your head even more.

Path 2 has plenty of processing power left over for guitar, so get creative! Or, patch an Electro-Harmonix Ravish Sitar stompbox in front of Helix to dive even deeper into the Indian ambiance.

Humbucker <> Single Coil Converters

Humbucker and single-coil pickups use different technologies, and sound inherently different. And of course, even pickups that use the same technology sound different. These converters aren't an exact emulation, but they do give the vibe—brighter for a single-coil (fig. 3.20), darker/warmer for a humbucker (fig. 3.21). The converters use a Low/High Shelf equalizer and two Parametric equalizers. Please note that these are optimized for the treble (bridge) pickup.

Figure 3.20 The **Hum to SC** preset converts a humbucker into having more of a single-coil sound.

Figure 3.21 The **SC to Hum** preset performs single-coil pickup to humbucker pickup conversion.

To emulate other pickup types, here's the approach I used.

1. Choose the guitar with the target sound (e.g., a Tele for a single-coil sound). Record about 30 seconds of strumming all the open strings. Strum evenly and consistently.
2. Record about 30 seconds of the source guitar whose sound you want to transform. It's important to mimic step 1's strumming style as closely as possible.
3. Follow each track with a spectrum analyzer plug-in (fig. 3.22). If your DAW doesn't include one, download the free SPAN plug-in from voxengo.com.

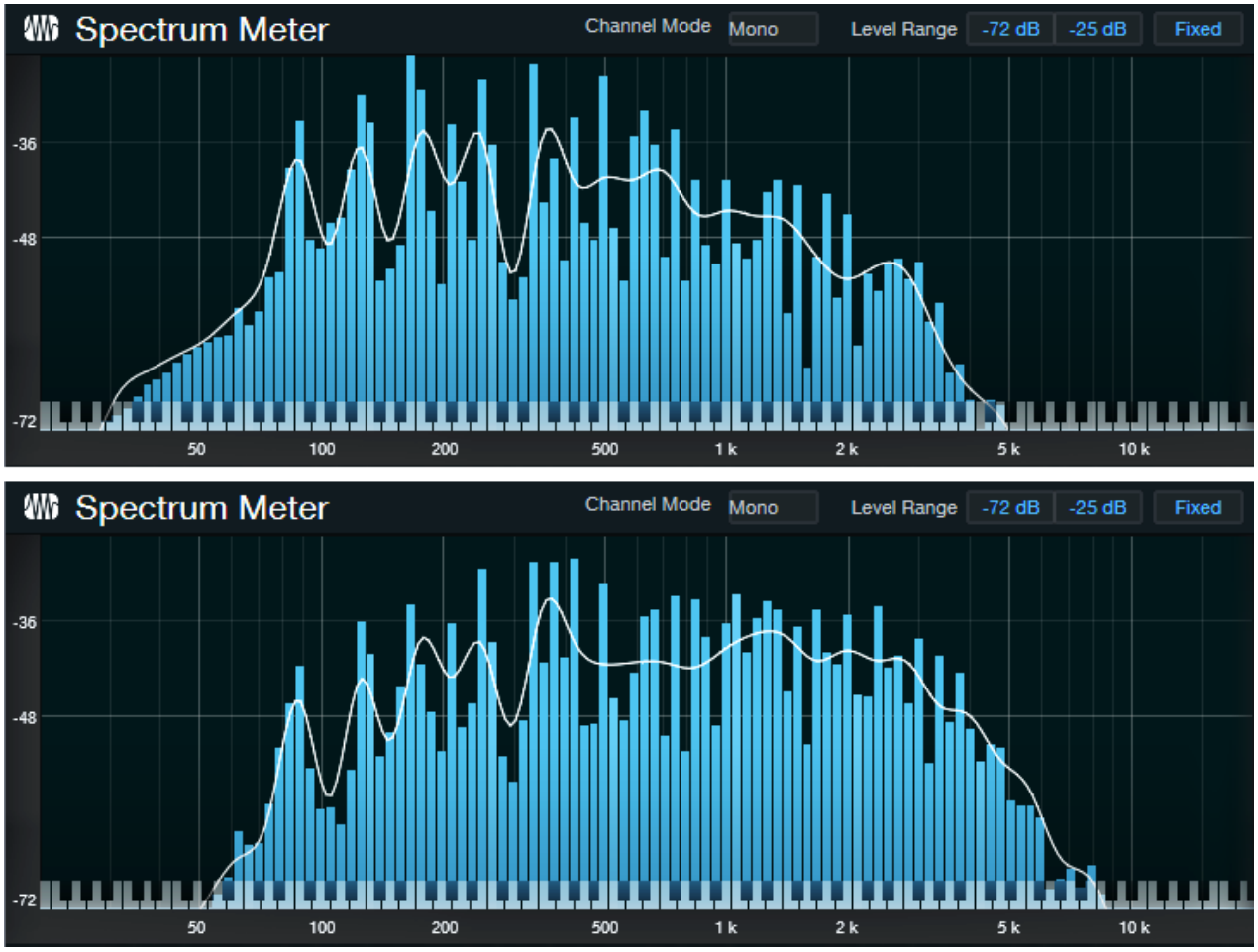


Figure 3.22 Frequency response spectrum for humbucker (top) and single-coil pickup (bottom). The single-coil pickup has an extended high-frequency response, and less low-frequency energy.

4. Insert at least one parametric EQ in the source guitar's track. Adjust the EQ settings so that the source guitar's frequency response mimics the target guitar's frequency response as closely as possible. You'll usually need more than one parametric stage, as in figs. 3.20 and 3.21. More stages give more detailed emulations. Sometimes a Low/High Shelf EQ can make broad bass and treble adjustments.

Fig. 3.23 shows the resulting humbucker pickup's response curve, derived from the EQ settings in fig. 3.20. The curve is more like a single-coil pickup than a humbucker.

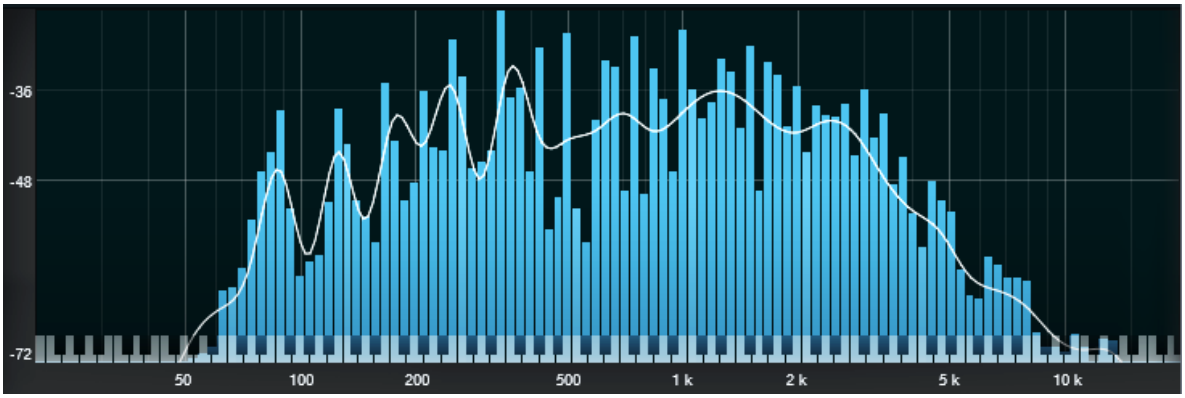


Figure 3.23 The resulting humbucker response more closely resembles a single-coil pickup.

Fig. 3.24, derived from the EQ settings in fig. 3.21, gives single-coil pickups a response that looks more like a humbucker. However, I decided to keep some of the high-end response. This brightens up the emulated humbucker a bit.

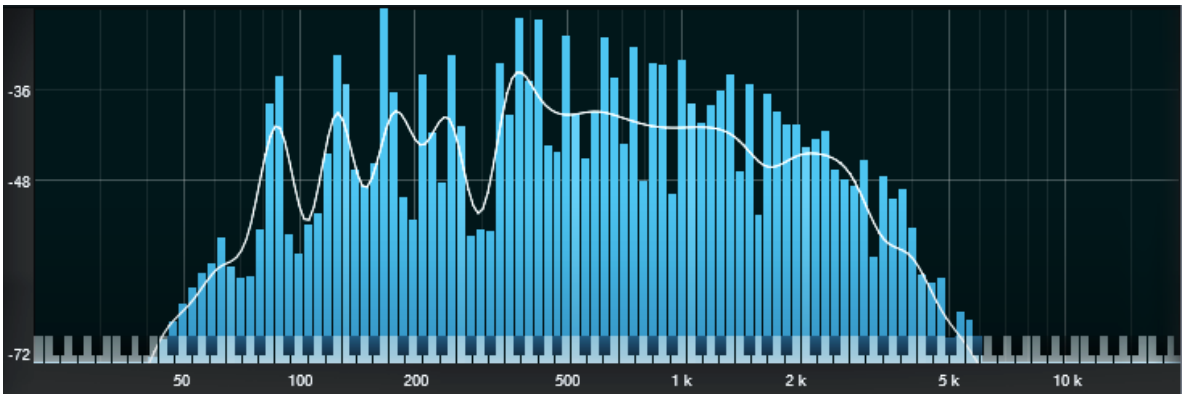


Figure 3.24 The single-coil response looks and sounds more like a humbucker.

While these transformed sounds give the feel of the pickups they're modeling, you don't always have to follow tradition. For example, when transforming a single-coil to a humbucker, note the distinctive "bump" around 2.3 kHz. Changing the gain at this frequency alters the character.

A bonus preset, **SC to SemiHollow**, converts a single-coil sound to the sound of a humbucker mounted in a semi-hollow-body guitar. However, it uses five EQ blocks. Detailed emulations require more parametric EQ stages, because the difference between sounds is often the result of multiple resonances.

But "close enough for rock and roll" applies here. The main point is you can coax sounds out of your guitar that aren't normally associated with a particular pickup technology.

Sci-Fi Feedback Effects Generator

Attention sound design fans: Helix can be coaxed into making extremely strange, feedback-based effects. Inserting the Send block later in a chain than the Return block lets you create a feedback loop that passes through external hardware effects. Or, simply patch the Return to the Send for Helix-only feedback. But first…

WARNING! RED ALERT!! This type of setup can generate uncontrolled feedback and extremely high levels. ***Always listen at very (very!) low levels, and never do these kinds of experiments with headphones on.*** A really good blast of high-level feedback could damage your hearing

Fig. 3.25 shows a setup that requires no external hardware—just patch the Send to the Return. This setup creates feedback around a Delay and 10 Band Graphic equalizer. Varying the EQ's sliders creates boosts that go in and out of feedback at different frequencies. The Delay adds an extra layer of insanity.

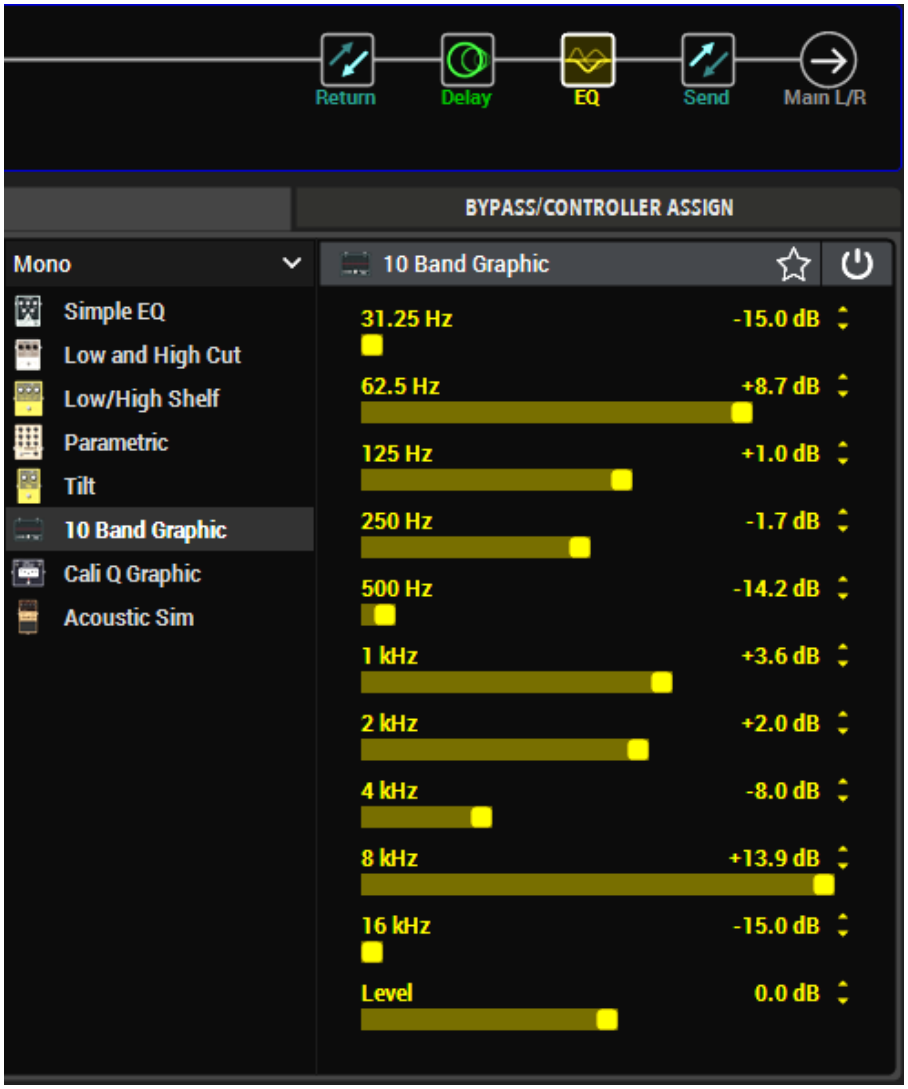


Figure 3.25 Yes, the Return is before the Send. This is on purpose.

This is just one possibility—placing reverb, modulation, and other effects inside a feedback loop can generate some pretty unusual sounds. So, if you need to take a break from playing guitar and score some sci-fi or horror movies, here's the ticket. Note that for keyboards, some of these sounds are worth sampling.

Deconstructed Guitar

Maybe you've seen food done as a deconstructed dish, where ingredients typically put together are served separately. For example, a deconstructed spring roll would plate all the ingredients inside it separately, with a dipping sauce on the side. You could wrap up the ingredients in whatever way you liked to construct your spring roll.

Wait—*what does this have to do with guitar?* Well, I had an opportunity to talk with Chris Jenkins, who won an Academy Award for the sound in "Mad Max – Fury Road" and worked on the documentary "The Beatles: Eight Days A Week—The Touring Years." He mentioned how for the surround mix of ancient Beatles concert footage, instead of upmixing, they deconstructed it by sending pre-equalized sound into special-purpose, acoustical spaces optimized for different instruments. Re-recording the "acoustified" sounds created the 5.1 mix. This is why footage from the 60s could sound natural in 5.1 surround.

Sound effects sometimes use a related approach. If they don't sound right when laid into a film's soundtrack, they might be played back in an acoustic space, re-recorded, and then added to the film. This makes them sound more "real world."

I prefer this deconstructed technique with stereo, dry guitar, but it also collapses well to mono (fig. 3.26).

Figure 3.26 Setup for "deconstructed" guitar sounds.

Split the guitar into a Dual Cab block (no effects or amps). Pan the cabs oppositely to “deconstruct” the guitar into the two cab sounds. Then, equalize the guitar sound as needed with the Parametric EQ. There’s no specific “desirable” curve. Apply a curve that sounds good with the cabs being used.

Try this technique with DI guitar as well as LCR mixes. Deconstructing the guitar opens up space in the center, and the guitar no longer has that flat “DI sound.”

Changing one of the cabs creates a different tone. In developing this tip, I tried a lot of cab combinations, and there were plenty of great-sounding options. Using Legacy cabs is convenient, because the Dual Cab block opens up with the Cabs panned hard left and right. They also have an interesting character. The version 3.50 Dual Cabs default with the two cabs panned to center, so to hear what they sound like in stereo, you need to re-pan every time you change cabs.

However, the version 3.50 cabs have more sophisticated miking and delay options, which adds considerable flexibility. One possible workflow is to open a Legacy Dual Cab, and find two cabs you like that have version 3.50 equivalents. Then, open a version 3.50 Dual Cab, load the cabs you like, pan them, and tweak the sound with the additional parameters.

Adding EQ extends the options further. For example, if a particular cab combination emphasizes the bass too much, trim the EQ’s low frequencies.

This may seem like an unusual technique, but it can add depth and interest to dry guitar parts. Also try it with acoustic guitar.

Truly Ghastly Sounds

Some people want ugly, nasty, disgusting, loathsome, repugnant, odious, vile sounds. Maybe they need to drive out unwanted house guests, have a Halloween gig coming up, were raised by wolves, or are just plain anti-social. Well, having had a few times when *I* needed to drive out unwanted house guests, I haven't forgotten about you!

This preset inserts two different amps in parallel, and inverts one path's polarity. The audio going through the amps is more similar than the distortion the amps add to the audio, so the distortion becomes more prominent.

1. Create a parallel path with two different amps, preferably ones with significant distortion.
2. In the Mixer block, choose Inverted for B Polarity (fig. 3.27).
3. Adjust *either* the A or B Path Level parameter for the thinnest sound that's also highly distorted. The object is to cancel out any audio that's *not* distortion.
4. Because so much of the signal cancels, you'll need to raise the output level. In the **NastyGhastly** preset, I got lucky—raising the Ch Vol parameters for the two amps sufficed. For more output, raise the Mixer Level.
5. If the sound isn't sufficiently awful, try different amp combinations, as well as editing the Drive and Bias parameters. Eventually, you'll find something that sounds sufficiently hideous.

Tip: For a slightly less horrific effect, use Amp+Cab blocks instead of amps.

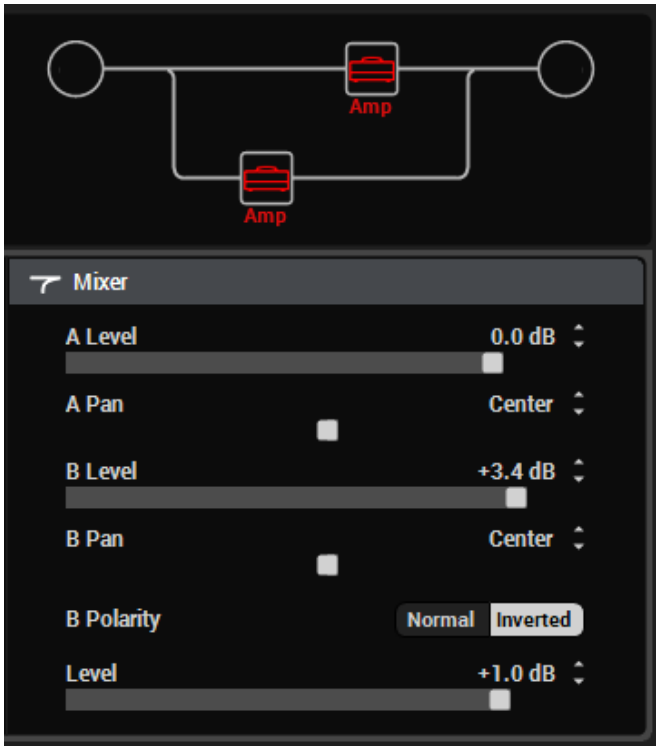


Figure 3.27 Don't say you weren't warned.

Sweet Soundscape

We can't leave this chapter with Truly Ghastly Sounds still ringing in your ears, so here's the antidote. This preset also answers the question "How do you use the Glitch Delay?" If you've had a tough day, plug in your guitar. Load up **Sweet Soundscape**, work the volume pedal to fade your guitar into the matrix of Glitch Delay and Particle Reverb (fig. 3.28), and enjoy tones that are the audio equivalent of audio chiffon cake.

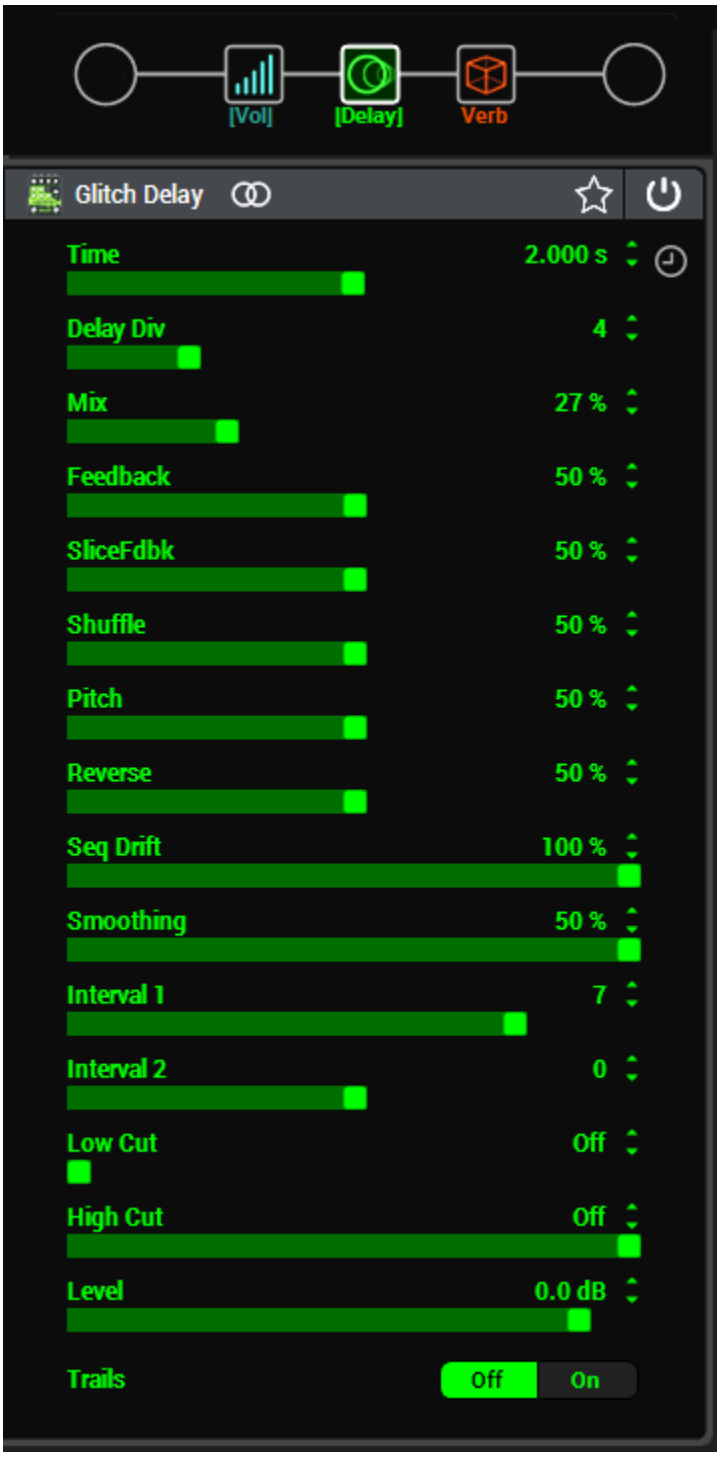


Figure 3.28 Glitch Delay settings for the Sweet Soundscape preset.

Chapter 4 | POLY BLOCK TECHNIQUES

The pitch-shifting block added in version 3.0 opens up exceptional possibilities. This chapter explores some of what you can do with this important, and powerful, block.

Version 3.0 introduced high-quality polyphonic pitch-shifting. As mentioned in Pitch/Synth Blocks, shifting pitch is a major technical challenge, and requires tradeoffs:

- Real-time pitch-shifting is *far* more difficult to implement than the non-real-time, offline processing done in DAWs and digital audio editors.
- Higher latency gives pitch-shifting algorithms more time to analyze the note pitches prior to transposing them. This improves pitch detection and tracking accuracy.
- Sound quality decreases as the transposition amount increases.
- Playing cleanly, especially with attacks, helps pitch-shifting algorithms recognize a note's pitch.
- High-quality, low-latency, real-time pitch transposition requires a lot of processing power.

The Poly block implements convincing, real-time pitch-shifting. Its quality even rivals some offline pitch shifters. However, note that the Poly block is mono-only. Stereo would require much more processing power.

The Poly algorithm is optimized for polyphony, so it's ideal for chords and strumming. Single-note lines don't need the DSP power that polyphonic shifting requires. The Simple and Dual Pitch processors, as well as the Twin Harmony, work well for single-note lines. They require less processing power and have lower latency. However, all Poly block-based processors include a Tracking parameter that trades off tracking accuracy for lower latency. This can optimize pitch-shift performance when combining single-note lines and chords,

Different Pitch-Shifting Flavors

The Poly processor has four different personalities:

- **Poly Pitch.** When enabled, the pitch slides from the non-transposed pitch (i.e., the effect's bypassed state) to a transposed pitch you specify. You can edit the slide's duration, as well as the time to slide back to the non-transposed pitch after bypassing Poly Pitch. (The sliding is like a synthesizer's polyphonic glide or portamento.) This block can produce steel guitar-like bending effects.
- **Poly Wham.** Use this for pedal-controlled pitch shifts. If your guitar doesn't have a vibrato tailpiece…now it does.
- **Poly Capo.** For simply transposing your guitar, polyphonic capo is the best choice.
- **12-String.** This gives the personality of a 12-string guitar, although I prefer to make my own 12-string presets based on the Poly Capo.

Common Parameters

Poly Pitch, Poly Wham, and Poly Capo have four common parameters:

- **Mix** varies the blend of the dry and processed sounds.
- **Level** sets the output.
- Transposing higher gives a brighter sound, while transposing lower gives a duller sound. **Auto EQ** compensates for timbral changes that happen when transposing pitch. With Auto EQ = 0, there's no compensation for timbral changes. Higher values apply more compensation (reduced brightness when transposing up, increased brightness when transposing down). For more information, see Poly Sustain.
- **Tracking** offers four choices, from X Fast (least latency and least accurate tracking precision) to X Stable (most latency and most accurate tracking precision).

Transposition with Poly Capo

In live performance, suppose a song modulates up a half-step during a solo, then modulates back down a half-step when the solo's over. Control the Interval parameter with an Expression Pedal. Set the Pedal's Min Interval to 0, and Max Interval to +1 (fig. 4.1). Push down on the pedal to modulate up a half-step. Pull back to return to the original key.

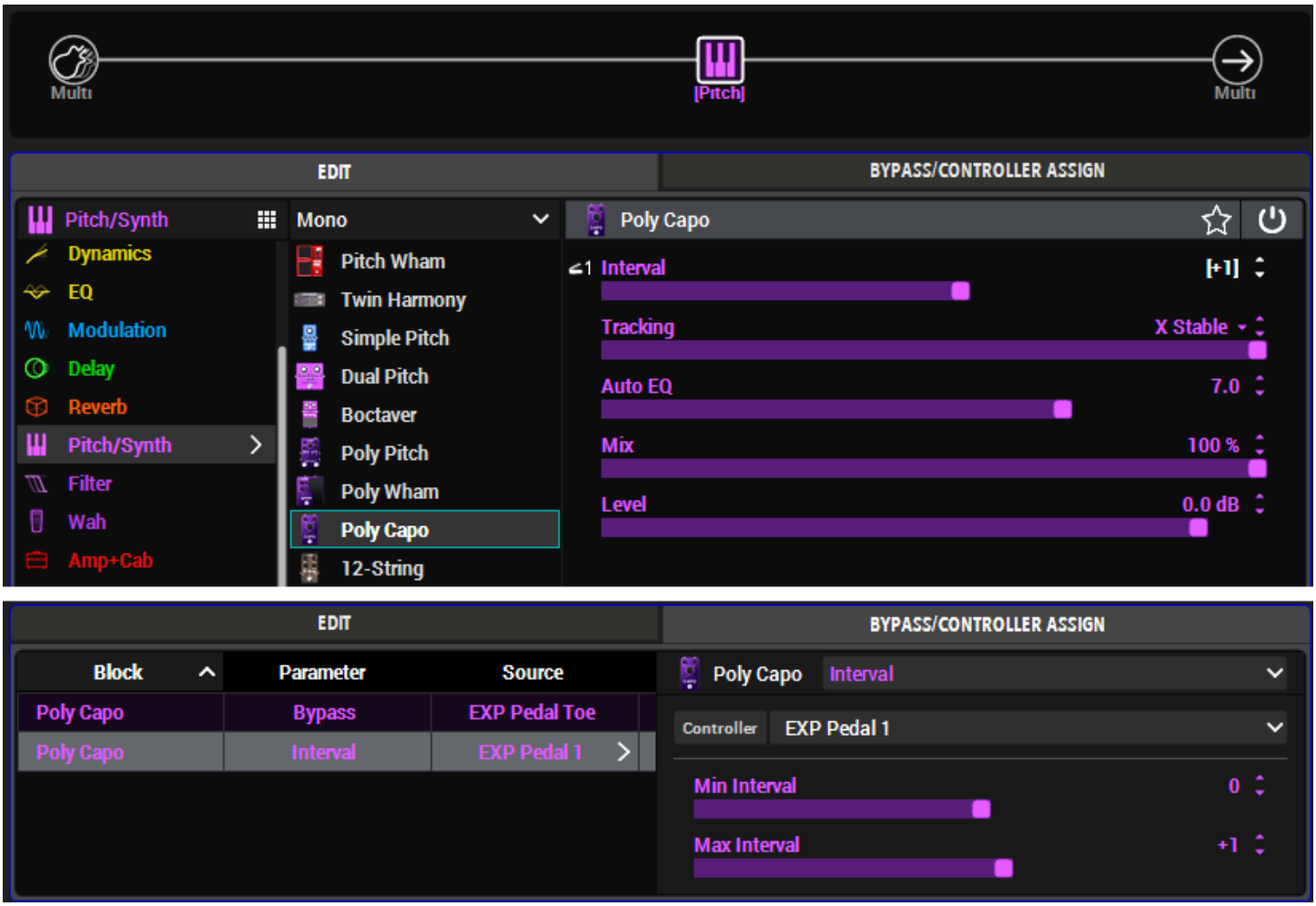


Figure 4.1 The Poly Capo can do real-time, pedal-controlled transposition to different keys. The top part of the image shows the Poly Capo settings. The bottom part shows the Expression Pedal settings for Helix Floor.

Transposition is also useful in the studio. When songwriting, use Poly Capo with Helix Native (or set up Helix/HX Stomp as an "outboard" plug-in) to assist in finding the best key for your voice. Record a rhythm guitar part. Use Poly Capo to change keys until you find one that's perfect for the song.

After finding the key, learn the song in the new key. However, the pitch transposition's sound quality is often good enough that you can simply use the transposed setting as is.

Tribute to Wes Montgomery

Wes Montgomery wasn't just another jazz guitarist—he created a unique, immediately identifiable style. There's no way you can duplicate the Wes Montgomery "sound" with a plug-in, because most of his sound came from his fingers. But we can pay tribute to that sound, with the preset **Tribute to Wes M** (fig. 4.2).

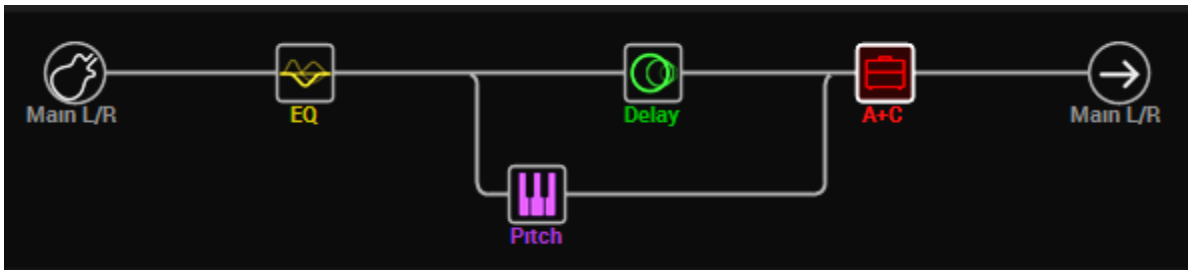


Figure 4.2 This preset emulates Wes Montgomery's single-note octave sound but includes a twist for extra realism.

He was famous for the way he played single-note octave lines, using his thumb to do downstrokes (and of course, didn't use an octave divider effect). His left hand muted the strings in between the octave strings. For example, if he played an octave on the first and fourth strings, the second and third strings were muted. Hitting the muted strings along with the octave strings on the downstroke created a fatter sound. He used this to advantage on his more "pop" recordings. But he also played block chord solos using upstrokes and downstrokes, often at blazing speeds, in jazz trios and quartets that *really* knew how to swing.

The Poly Capo block provides the octave-below transposition. The "twist" for this preset is the delay on the main, non-transposed pitch. A downstroke hits the higher string a few milliseconds later, so the Simple Delay is set for 15 ms (or use 20 ms for a more pronounced effect). This produces a sound that's closer to Wes Montgomery's playing. Bypass the Simple Delay, and you'll hear the difference.

His main axe was an L-5, which was the first guitar to use f-holes. Built in a conceptually similar way to cellos, it had a big, booming sound. Maybe there's an L-5 body Impulse Response out there somewhere… meanwhile, the EQ at the beginning simulates the warmer sound of his heavy-gauge (0.014 for the high E) flatwound strings. He didn't roll off his tone control, because the strings themselves weren't that bright. For this preset, use the neck pickup.

He reportedly used a Fender Super Reverb in his early years, then a Standel Super Custom, and finally, a Twin Reverb. So, this preset uses the US Double Nrm Amp+Cab model, with the drive turned down.

Low Tuning Presets

Low tunings tune down guitar strings compared to standard tuning. This gives a deep, fat timbre that's common in metal. Typically, low tunings use thicker strings (e.g., at least a 0.011 for the top string—thinner strings aren't under enough tension to prevent the pitch from varying as the string decays).

The Poly Capo creates low tunings electronically. For example, transposing down 3 semitones is like tuning your strings to (low to high) C# F# B E G# C#. The downsides of transposing with the Poly Capo are:

- The guitar strings will not be at the same pitch as what you hear from your amp. For live performance, the amp sound will probably mask the string sound. In the studio, around-the-ear closed headphones can isolate you from the string sound.
- You'll need to play the song in a different key. If a song was written in A and you transpose down 3 semitones, you'll need to play it in C.

This technique can also enhance layered guitar parts. A low tuning changes the guitar's timbre, so it sounds different from other layer(s).

Standard Low Tuning

The **LowTuning** preset (fig. 4.3) tunes your guitar down -5 semitones, which bulks up the tone.

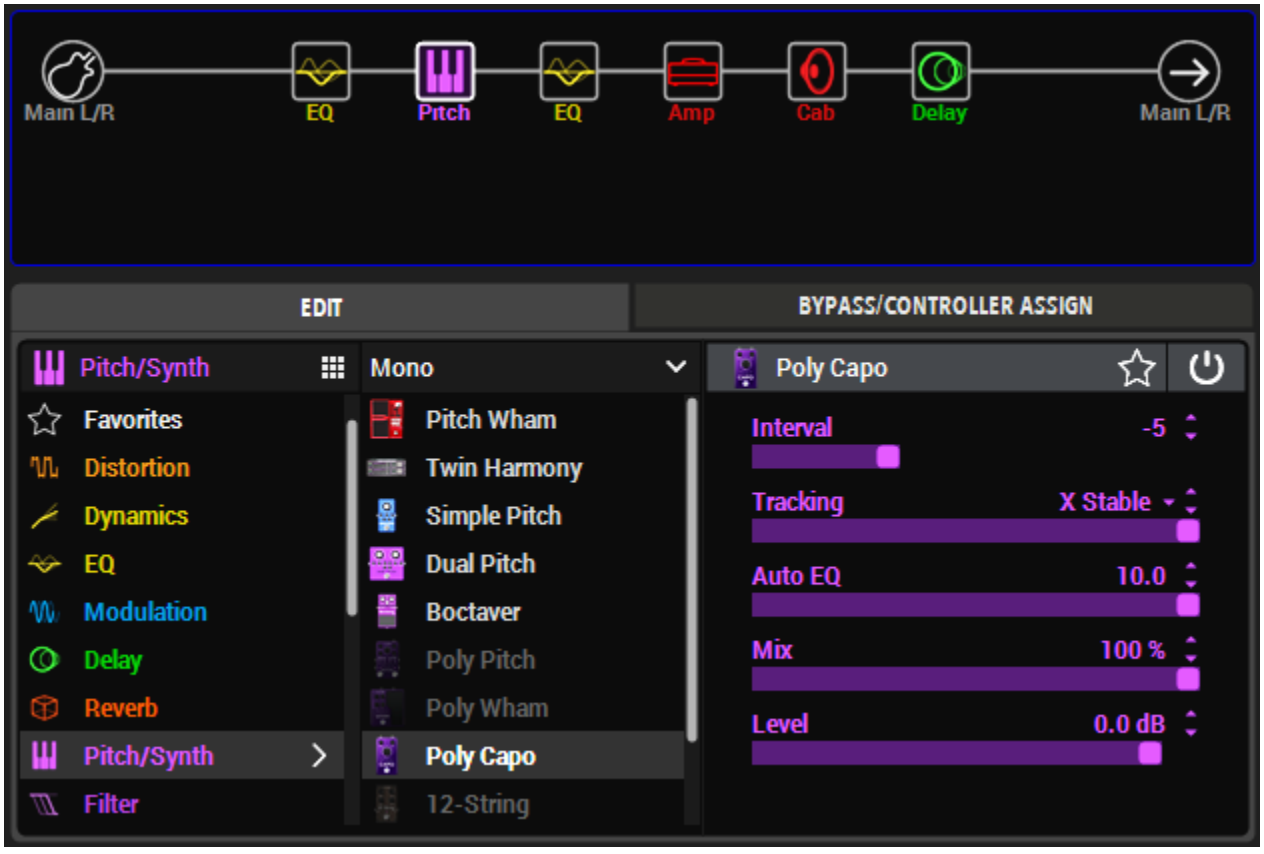


Figure 4.3 Low tuning simulation with the Poly Capo.

The previous edition of this book had to work around DSP issues. With version 3.50 Dual Cabs, this is a better-sounding, more straightforward preset. The output Mixer can pan the two cabs for stereo imaging. A stereo Simple Delay provides a stereo output that collapses well to mono. However, deleting or bypassing the Simple Delay for mono outputs may sound slightly better. There is no DSP left over to add other effects.

Customizing

- This preset uses the Das Benzin Mega amp, which requires significant DSP power. Using a less power-hungry amp allows adding more effects, including post-cab stereo effects.
- 38 amps are not available with this preset.
- Deleting the Low and High Cut EQ before the Pitch block allows using all but 19 amps. It also opens more options for the post-cab stereo effect.
- Deleting the stereo Simple Delay allows using all the amps except for the Brit Plexi Jmp, PV VitriolCrunch, and PV Vitriol Lead.
- Deleting the Low and High Cut EQ and Simple Delay allows using all the amps except for the PV VitriolCrunch and PV Vitriol Lead.

Bi-Amped Low Tuning

Thanks to version 3.50's efficient cabs, bi-amping can include more effects with low tunings (fig. 4.4). There's more about bi-amping in the next chapter.

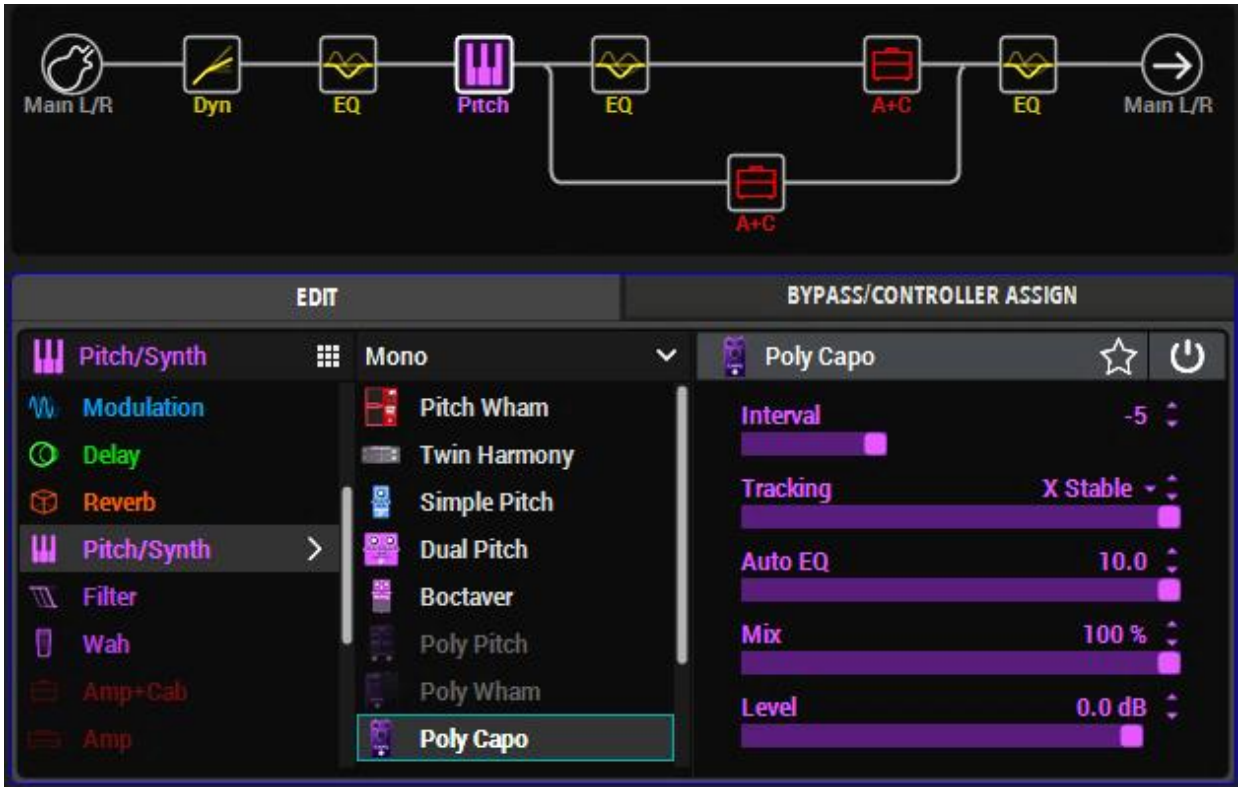


Figure 4.4 The **BiAmpLowTuning** preset allows stereo bi-amping.

The choice of amps is limited when including all these effects. This preset uses a US Small Tweed for the highs, and a Soup Pro for the lows. The High and Low Cut EQ block before the Poly Capo conditions the signal going to the Poly Capo. The Simple EQ in the high-frequency path conditions the signal going to the US Small Tweed, and the stereo Simple EQ at the output acts like a tone stack that preserves the Mixer's panning.

Poly Pitch

This effect can give pseudo-steel guitar effects. The guitar slides from one pitch to another over a specified time, and then slides back down again. Poly Pitch is also useful when you need to play with out-of-tune instruments, like a piano whose tuning changed due to temperature variations.

Poly Pitch doesn't do anything unless you enable or bypass it. When enabled, the pitch slides up based on the Interval parameter (fig. 4.5). For example, if Interval is +12 and you enable the effect, the pitch changes to an octave higher. It's like hitting a chord on a steel guitar, and then moving the bar up an octave.

The slide's duration depends on the Shift Time parameter (0.0 to 8.00 seconds). For example, a 500 ms setting would take half a second to transpose from the original pitch to the new one.

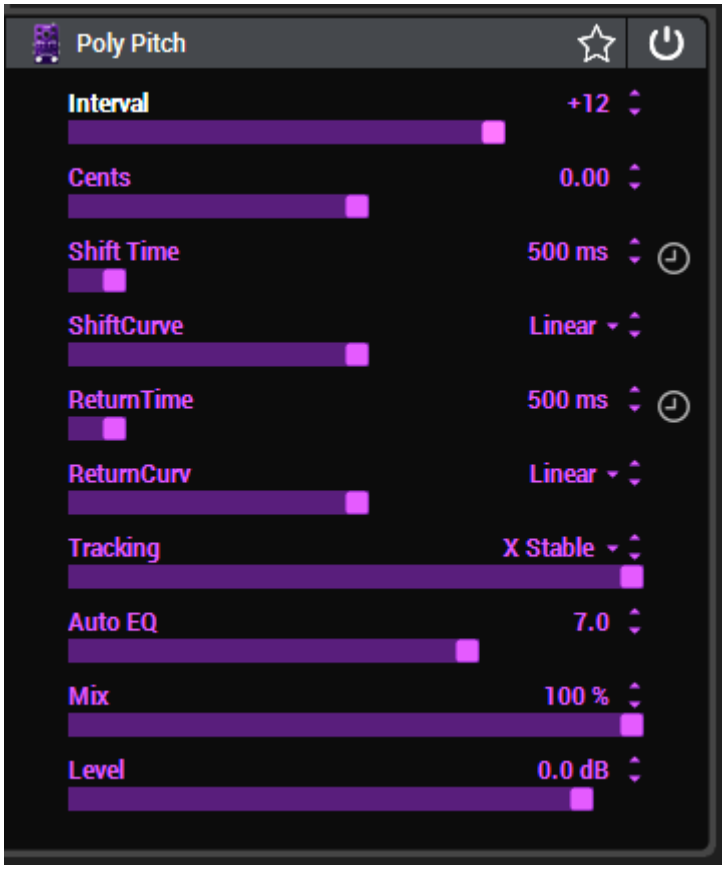


Figure 4.5 Poly Pitch can give pseudo-steel guitar effects.

Bypassing the Poly Pitch triggers the Return Time. This sets the time for the transposed pitch to return to the original pitch.

The Shift Curve Parameter

This parameter controls the rate of the Shift and Return times. The Linear setting changes pitch at a constant rate. The five StartSlow options change pitch slowly at first, but then the pitch change speeds up. The five StartFast options do the reverse—the pitch change starts rapidly at first, but then the rate of change slows down. Fig. 4.6 summarizes how this parameter affects pitch changes.

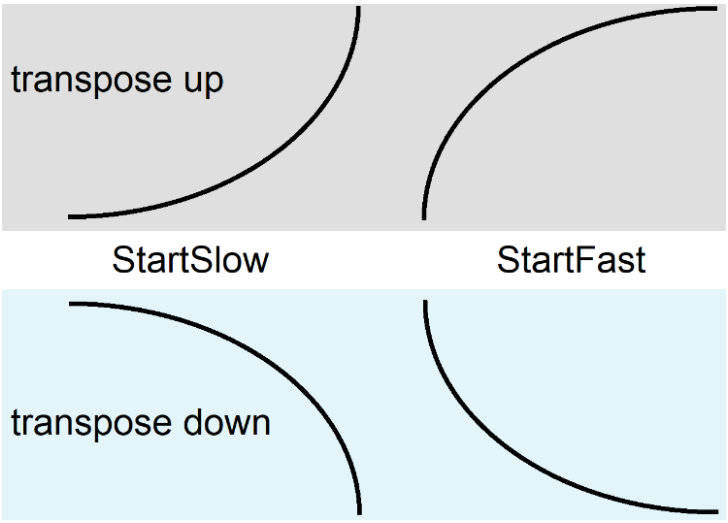


Figure 4.6 Typical StartSlow and StartFast curves. The upper curves show the rate of change when transposing upward, while the lower curves show the rate of change when transposing downward.

Pitch Fixes

Sometimes in the studio, you need to tune your guitar to a track that's not at concert pitch. For example, a piano might be just a little sharp or flat, but everyone tuned to it, and now you're expected to do an overdub. It can be time-consuming to keep re-tuning all six strings to match the out-of-tune track. A faster option is to tune the guitar correctly to itself, and then process it through the Poly Pitch. By *not* sliding the Poly Pitch, changing the Cents parameter changes the pitch of all six strings simultaneously (fig. 4.7). Simply play along and adjust the Cents parameter until the guitar is in tune.

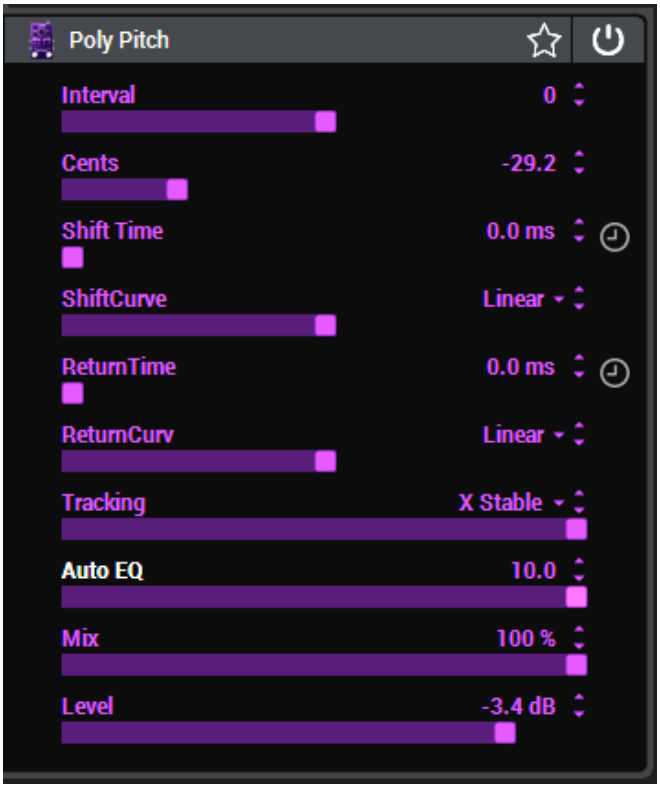


Figure 4.7 Setting Shift Time to 0.0 and Mix to 100% allows fine-tuning your guitar's pitch to match an out-of-tune track.

Nashville Tuning

Nashville tuning isn't an alternate tuning in the usual sense, because the strings still tune to E A D G B E. Instead, Nashville tuning retains a guitar's 1st and 2nd strings, but uses the 3rd through 6th octave-higher strings from a 12-string set (or a Nashville tuning string set, such as those from D'Addario and GHS).

This tuning delivers a bright, "chimey" guitar sound typically heard on country sessions, but also on rock recordings. You can hear Nashville tuning on:

- "Hey You" (Pink Floyd)
- "Dust in the Wind" (Kansas)
- "Wild Horses" (Rolling Stones)
- "Wicked Game" (Chris Isaak)
- "Gimme Danger" (Iggy Pop)

Most guitarists dedicate a guitar to Nashville tuning to avoid constant string changes. Furthermore, changing string tension requires adjusting the intonation, and possibly the truss rod. Some guitarists dedicate a 12-string guitar to Nashville tuning, because the nut is cut appropriately for the octave-higher strings.

The **NashvilleTuning** preset (fig. 4.8) isn't identical to Nashville tuning, because it transposes all six strings. So, it can't retain the standard pitch of the 1st and 2nd strings. However, it fulfills a similar role when blending in with a mix.

With electric guitar, I recommend using the neck pickup. Note that there's not enough DSP power to add a guitar amp or Amp+Cab.

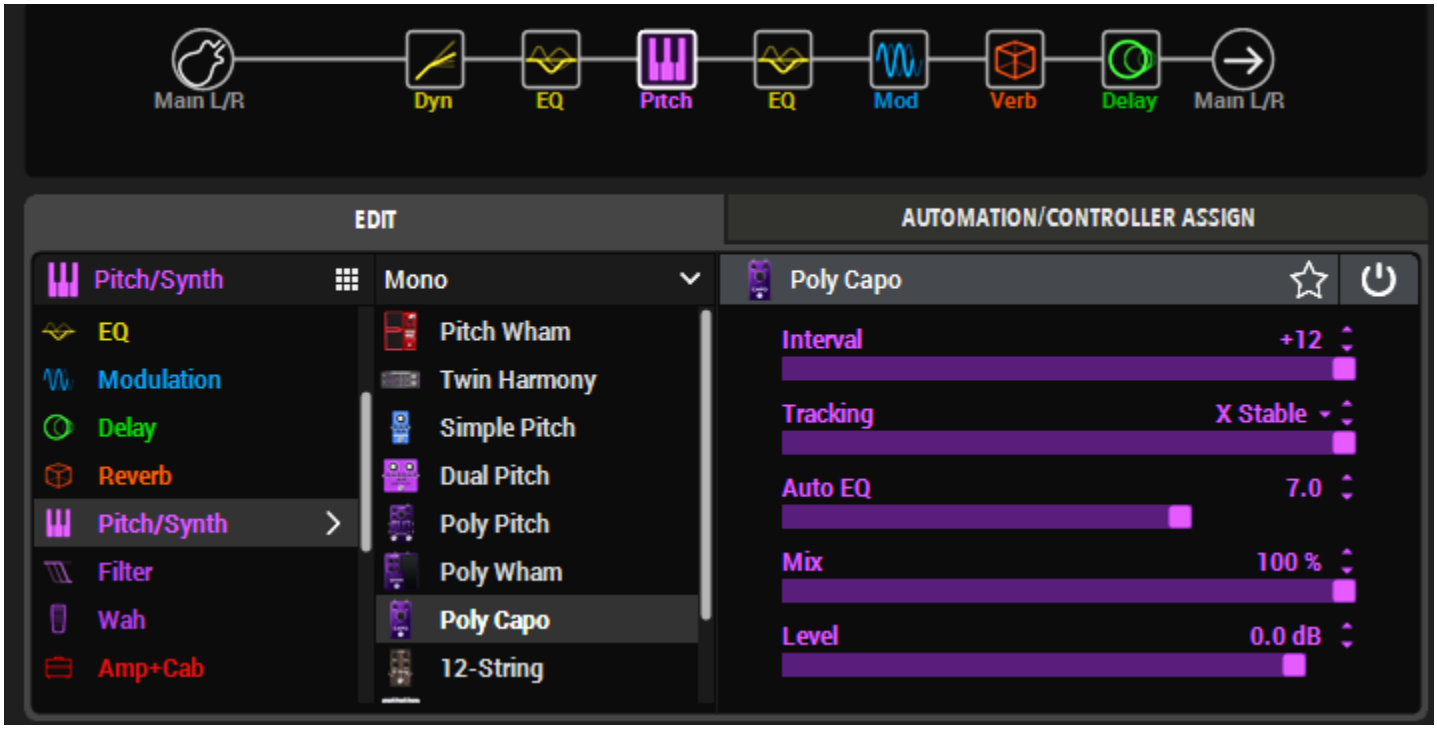


Figure 4.8 **NashvilleTuning** preset.

Customizing

- **LA Studio Comp.** The Poly Capo likes a clean audio signal (and relatively clean playing). This block uses Limit mode, instead of Compress, to tame peaks going into the rest of the chain.
- **Parametric EQ.** Inserting this before the Poly Capo offers several options:

    - To improve the Poly Capo's stability, reduce the low frequencies with the Low Gain and Low Cut, and trim the high frequencies with the High Cut.
    - For a beefier sound, don't cut the lows quite as much.
    - To brighten the sound, add a substantial upper Mid boost.
    - For a warmer tone, reduce the midrange response.

- **Poly Capo.** Because Nashville tuning is designed more for chords, set Tracking to X Stable because speed isn't as important as it would be for a blazing fast solo. Interval is +12. For a completely different effect, set Interval to -12 and play chords higher up on the neck. With this technique, the 6th string sounds like it's adding a bass note.
- **Auto EQ** compensates for the brighter tone that occurs when transposing audio higher. Increase Auto EQ for a slightly warmer sound. Lower it for a brighter sound.
- **Layering.** To emulate layering a standard guitar with a Nashville-tuned guitar, change the Pitch block's Mix parameter from 100% to a blend of the two sounds. However, also look at the LayerdNashLowCPU preset for more information on how to do this effectively.
- **Low/High Shelf.** Unlike a true Nashville-strung guitar, using the Poly Capo transposes the 1st and 2nd strings up an octave, along with the other strings. You may like the results, but for a more traditional sound, lower the post-Pitch block EQ's High Gain control. The Low Gain's -4.0 dB setting reduces lower midrange frequencies, which works well when layering the Nashville-tuned guitar with a conventionally tuned guitar. However, when playing the Nashville-tuned guitar by itself, consider raising this setting to increase the lower mids.
- **Chorus Modulation.** This adds shimmer. It modifies the default setting with a slower Speed, less Predelay time, and a Sine waveshape. The preset loads with the Chorus bypassed.
- **Reverb.** This uses the default Plate reverb, but with less Decay time, and a lower High Cut frequency.
- **Dual Delay.** See Dual Delay-Based Mono-to-Stereo Conversion in Chapter 11 for information on how this provides a rich stereo image. Modulation Mode is enabled and set to Chorus for a very subtle shimmer. This preset collapses well to mono, so you can keep the Delay enabled in stereo or mono setups. Delete or bypass this block if you'll be using only mono setups.

Layered Nashville

The **LayeredNashville** preset (fig. 4.9) uses Path B (with an LA Studio Comp) as a parallel path for dry guitar. The blocks are handled similarly to the **NashvilleTuning** preset; however, the Mixer allows for variations. For example, you can pan the dry and Nashville-tuned guitars differently, set a balance of the two, or control one path's level with an Expression Pedal.

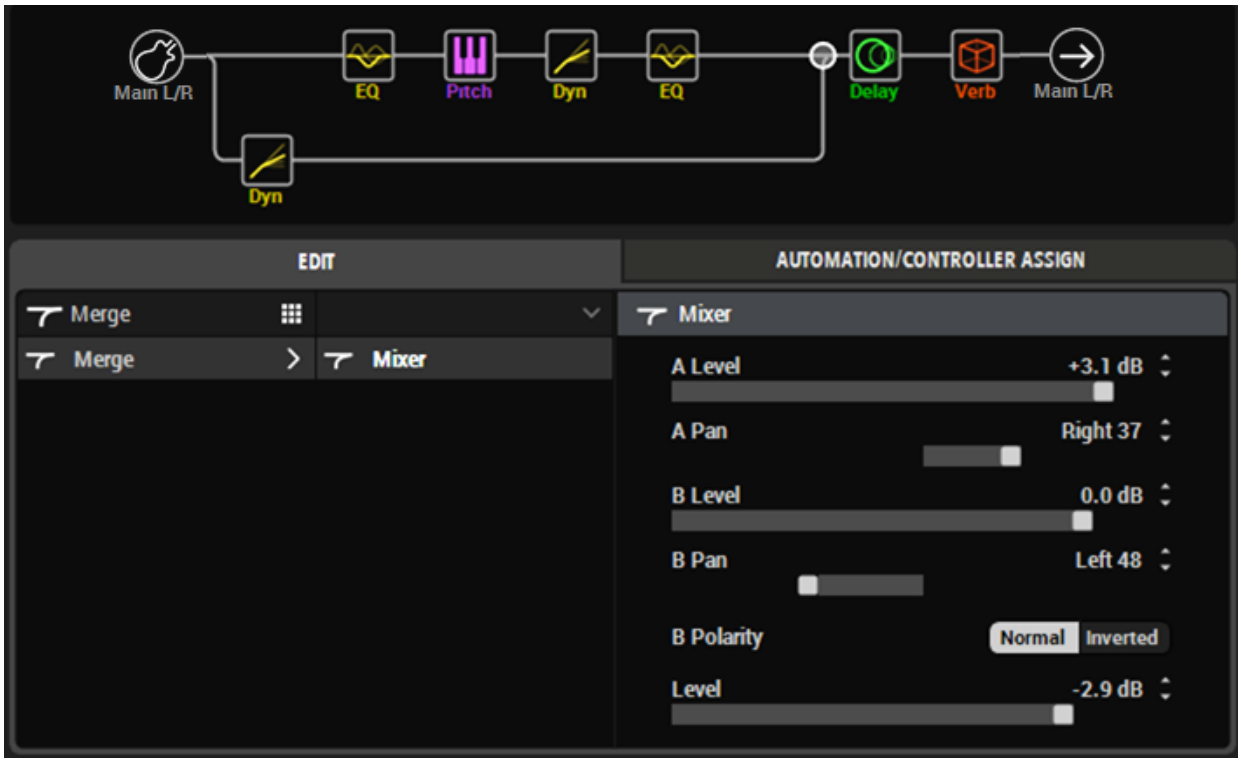


Figure 4.9 This preset layers a dry sound with the Nashville-tuned sound.

There's enough processing power left over to add another block to Path B's dry guitar sound. All the mono EQ, Wah, and single/Dual Cabs will be available, as well as a lesser number of Modulation, Delay, legacy Reverbs, and Filters (but no guitar amps).

Customization

The output Mixer can add to the illusion of layering:

- Pan the fundamental and octave sounds oppositely to differentiate them.
- Balance the two layered sounds by altering their levels.
- For B Polarity, choosing Inverted instead of Normal can sometimes give a fuller sound. This doesn't cause cancellation when using the preset in mono, because the two paths don't have much in common.

Layered Nash+Amp

LayeredNash+Amp removes the Dynamics processors and Reverb from the previous preset. This frees up enough processing power to access all but 19 Amp+Cabs (fig. 4.10), assuming you're using the new version 3.50 cabs. To be able to choose all Amp+Cab models except for the PV VitriolCrunch and PV Vitriol Lead, remove the Delay. This trades off stereo imaging (although the Mixer can still pan the two paths).

Figure 4.10 Removing some of the processors allows adding an Amp+Cab in Path B for the non-transposed guitar.

LayeredNashLowCPU

In the previous examples of layered Nashville tuning, dedicating a path to the dry guitar allowed optimizing the other path for the Poly Capo, as well as including a processor for the dry guitar. The tradeoff is being limited to effects that don't need lots of processing power.

The **LayeredNashLowCPU** preset doesn't have a separate guitar path (fig. 4.11), so the Poly Capo's Mix control sets the blend of dry guitar with the octave-above effect. In return, you can use any of the stereo or Legacy reverbs, and most of the stereo Delays. The rich, bright, smooth sound has a spacious stereo image.

Figure 4.11 This preset frees up enough processing power for complex stereo effects.

Layered Nash More FX

The **LayerdNashMoreFX** preset (fig. 4.12) takes a different approach to adding more effects. As with the **LayeredNashLowCPU** preset, it layers dry guitar with the Poly Capo by setting the Poly Capo's Mix parameter to around 50%.

Figure 4.12 This preset provides dry and octave above sounds yet frees up enough processing power to add several stereo effects.

After the audio leaves the Poly Capo, there's EQ to add brightness, followed by the Gray Flanger. This is a processing-heavy modulation effect. An LA Studio Comp, in Limit mode, tames the characteristic peaks caused by flanging. This preset also emphasizes stereo. Double Take, the Reverb, and the Simple EQ at the output are all stereo. There's not enough processing power left over for a Spring reverb, but most of the other Legacy reverbs work.

Gourmet 12-String

The **Gourmet12-String** preset (fig. 4.13) wraps processors around the Poly Capo to produce a rich, bright, big 12-string sound for stereo or mono. It's optimized for the bridge pickup, but other pickup positions work too.

This is based on the **LayeredNashville** preset. However, the sound is more like one guitar than layered guitars, because the second EQ (Low/High Shelf) doesn't emphasize the highs as much. Also, to give a 12-string's characteristic shimmer, the Dual Delay toward the output adds subtle modulation.

The crucial difference is the Simple Delay that precedes the Poly Capo. With a physical 12-string, there's a slight delay (around 20-25 ms) between hitting the main string and its associated octave string. The Simple Delay emulates this effect. Your ears identify this as part of a 12-string's characteristic sound.

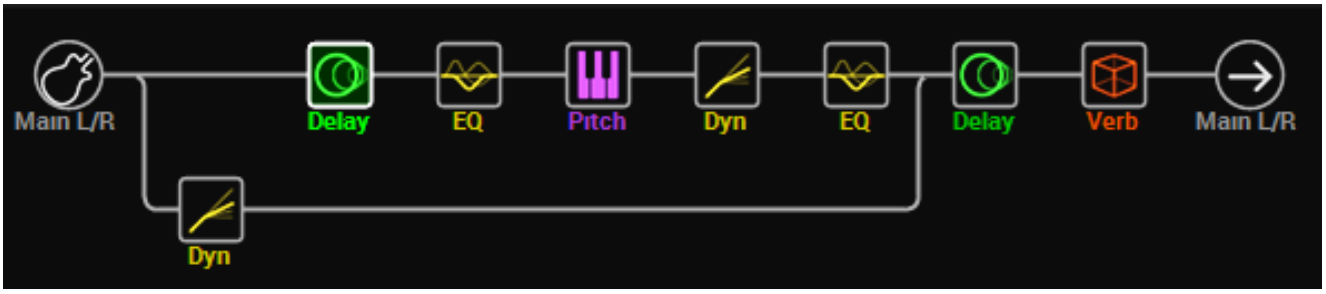


Figure 4.13 For a synthetic 12-string sound from a digital effects processor, this preset comes very close.

For live performance, only some blocks are suitable for footswitching. All the blocks before the Mixer are essential parts of the 12-string sound. The post-Mixer Delay and Reverb are good candidates for footswitching. One (or both) compressors can also benefit from footswitching. Then you can call up sounds that are more or less compressed.

Poly Block Distortion Stack

You don't always need an amp, which can save DSP power. The **PolyDistortStack** preset (fig. 4.14) has plenty of effects, because any of several distortions (in this case Stupor OD) can provide the "amp" sound. The cabs adding the necessary realism. This allows inserting the LA Studio Comp dynamics, Poly Capo, stereo Simple Delay for slapback echo, stereo Double Tank reverb, and a final stereo 10 Band Graphic EQ at the output as an overall tone control.

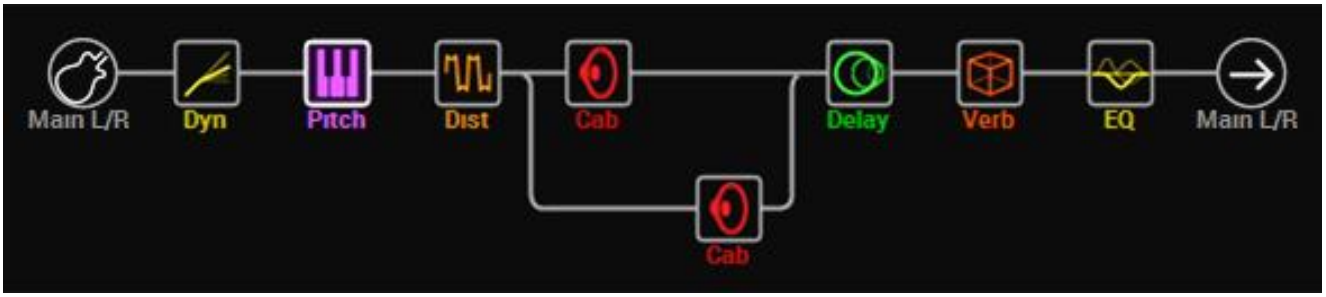


Figure 4.14 Presets based on Distortion+Cab can give a welcome alternative to conventional Amp+Cab presets.

The preset opens with the Poly Capo tuned up a fifth (7 semitones), with 50% dry/wet mix. Play a full barre chord—it's fun.

Bass Blast

Bass Blast is a “necessity is the mother of invention” preset. I was putting together a solo act for a local venue, with the guitar mostly playing chords to accompany vocals. But I really, *really* wanted a bass rumbling along with the chords. This preset parallels a bass behind the chords, as if a phantom bass player was doubling the chords’ root notes on bass. The Poly Capo block provides the octave lower notes. However, I didn’t want the octave effect on higher notes. So, this preset’s goal is to isolate only the guitar’s lower notes and transpose them down an octave.

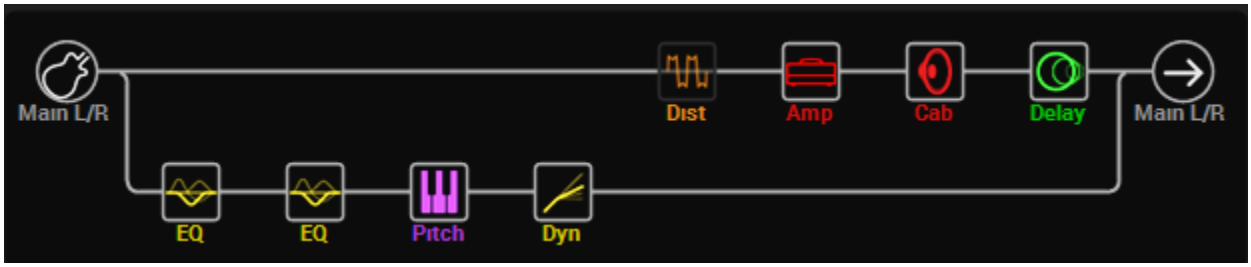


Figure 4.15 Preceding the Poly Capo with two filters passes only the guitar’s low notes.

Adding two EQs before the Poly Capo (fig. 4.15) helps keep the higher notes from reaching the Poly Pitch block. The first EQ cuts out everything except the low notes, and uses a 10 Band Graphic EQ. The second EQ boosts the low note levels with a Parametric EQ. As a result, you hear only the guitar’s *lower* strings play an octave lower. The compressor that follows the Pitch block helps bring up the sustain, which makes the low frequencies sound more like a bass.

The level going into the Poly Capo is relatively low, so its Level adds +6.0 dB. The Mixer adjusts the balance between the amp path and the bass path. You may want less bass level than the preset’s default setting.

Customization

The playing style that works best is barre chords, where you hit the lowest string for the bass sound. You may get so used to this sound that if you remove the parallel path, you’ll wonder what happened to your huge, powerful guitar! In any case, try to play cleanly, and don’t hit several of the low strings at once—it will sound as muddy as playing several bass strings simultaneously.

The customizations will vary, depending on how you to plan to use this preset. As you make changes, though, you’ll almost certainly need to adjust the two path levels in the output Mixer.

- This preset requires significant DSP power. With the Group 1 and 2 Amps (as described in the section Amp+Cab Blocks and Processing Power), you can follow the amp with a version 3.50 Dual Cab. Group 3 Amps are eligible too, except for the Voltage Queen, Mandarin 80, Cali IV Rhythm 1, Cali IV Rhythm 2, and PV Panama.
- Adding effects to the guitar path depends on the Amp+Cab. The US Small Tweed used in the preset allows for a stereo Simple Delay at the output, and the Heir Apparent distortion (it’s bypassed when you load the preset). Or, replace the distortion with EQ or Dynamics, or don’t use any effect in front of the amp so you can add another post-amp effect (e.g., Modulation, or a Legacy Reverb).
- For live use, you’ll probably want to be able to mix the bass path in and out. To do this, assign an Expression Pedal to the Poly Capo Level parameter.

BassBlast 4-Path

The **BassBlast 4Path** preset upgrades the **Bass Blast** preset for 4-Path Helix family processors (fig. 4.16). The bass path remains the same, but the extra processing power allows a more evolved guitar path. The sonic impact is pretty impressive, and can help fill out the sound of a live performance act based on only guitar, voice, and percussion.

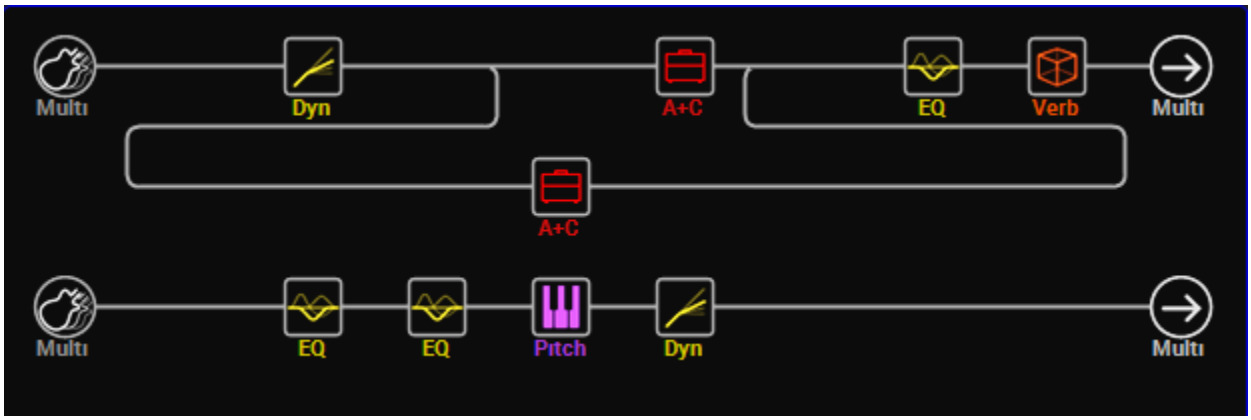


Figure 4.16 This preset enhances the Bass Blast preset for 4-Path Helix devices.

Still, the amount of processing power isn't infinite. So, the effect options you can choose depend on the Amp+Cab chosen for the guitar path. This preset uses bi-amping, and the Mixer spreads the amp sounds in stereo. As a result, the EQ must be stereo as well. But you don't have to get this fancy—a single Amp+Cab will free up processing power for other effects.

Chapter 5 | BI-AMPING TECHNIQUES

Bi-amping a guitar amp has the same benefits as bi-amping studio monitors—you can optimize the amps to perform best with specific frequency ranges.

Bi-Amping in a Parallel Path

Bi-amping splits incoming audio into parallel, high- and low-frequency bands. Like speakers that have high- and low-frequency drivers, bi-amping can optimize audio for each band. Useful applications include:

- Distort high and low frequencies independently for a more defined sound.
- Send only the highs through reverb for a bright sound. Leave the lower frequencies unprocessed.
- Use different delay times for the high and low frequencies (e.g., set the high-frequency delay for twice the lower delay's speed).
- Chorus, pan, modulate, or otherwise process the high and low frequencies independently for more interest and variety.

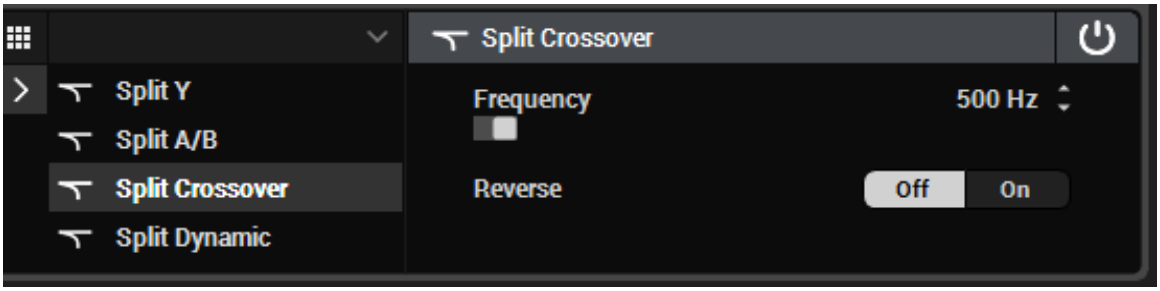
With amps, please note the following limitations with bi-amping in HX Stomp, or a single 4-Path Helix path:

- The PV VitriolCrunch Amp+Cab and PV Vitriol Lead Amp+Cab require too much DSP power to run in parallel. Nor can you run just the Amps in parallel and mix them to a single Cab.
- Any other combination of two Amp+Cabs will work. With amps that don't need a lot of DSP power, you'll be able to add effects.
- Several workarounds can conserve DSP processing power, like two amps sharing a single cab that's inserted after the Mixer (as described later).
- There's less level going into the amp because fewer frequencies are feeding it. You may need to increase Drive to compensate.

Next, let's explore several ways to split the guitar into two bands.

Input Split Block

As covered in Chapter 2, the Input Split includes a Crossover that can split incoming audio into high- and low-frequency bands. This is the simplest way to create a bi-amped setup:



- The **Frequency** parameter sets the Crossover point, between 25 Hz and 15 kHz.
- Frequencies *above* the crossover point go to Path A.
- Frequencies *below* the crossover point go to Path B.
- The **Reverse** parameter reverses the Paths. Choosing this sends frequencies *above* the crossover to Path B, and frequencies *below* the crossover to Path A.

The filter slopes are 24 dB/octave, which are moderately steep (fig. 5.1). This provides good isolation between the high and low frequencies. For amps, a steep slope is usually best. Note that the low frequency audio fades out above the crossover point, while the high frequency audio starts fading in before the crossover point. So, there is some overlap between the upper and lower frequency ranges. However, when added together, the frequency response is flat.

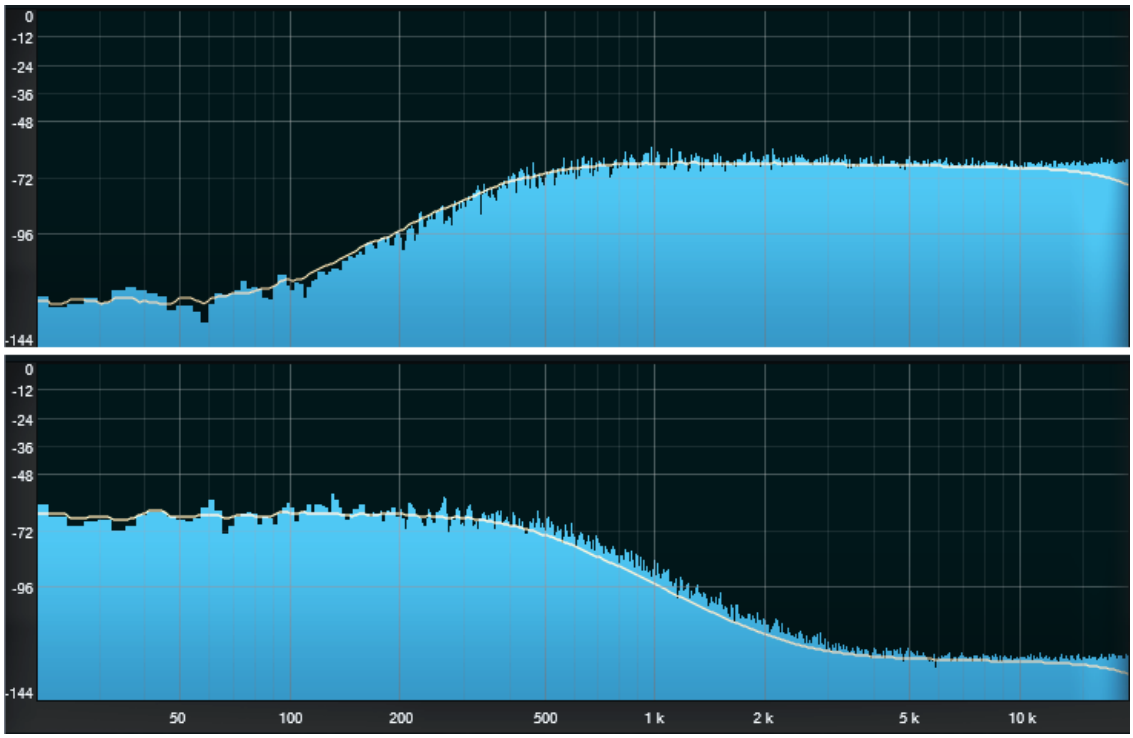


Figure 5.1 Input Split block filter responses for the high- and low-frequency splits. The crossover frequency is 500 Hz. The filter slopes are 24 dB/octave.

Shelf Split

The Input Split filter's slopes create a distinct separation between the high and low frequencies. However, you might not want as much separation with some effects, like modulation or delay (or even with some amps, particularly clean ones). Splitting with Shelf filters creates a subtle, approximately 6 dB slope over each band's full frequency range (fig. 5.2).

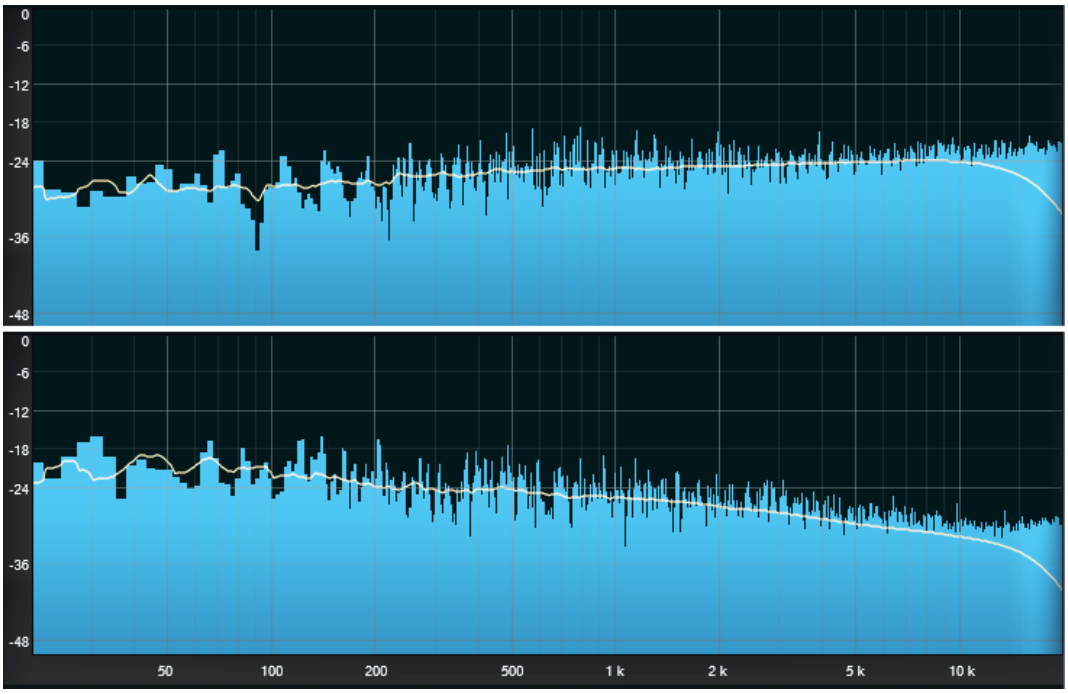


Figure 5.2 Using Shelf EQ to split high and low frequencies creates more overlap between the two bands.

This method requires adding an extra block in each split to provide the filtering (fig. 5.3).

Figure 5.3 The Shelf settings create low and high-frequency splits. In this preset, the high and low bands are going through modulation effects.

The Shelf frequencies are an initial starting point. The slope changes as you change the frequencies. For a steeper slope, increase the Low Gain and High Gain in their respective EQs. However, with steeper slopes, summing the two filters together no longer gives a flat response. Instead, there's a broad midrange scoop, with high and low frequency boosts. Then again, that may be what you want—experiment!

3-Band Comp Bi-Amp Split

Defeating the 3-Band Comp's compression transforms it into a 3-band crossover. Using two of those bands can do bi-amping (fig. 5.4):

- Because a 0.0 dB Threshold for all bands defeats compression, the Ratio, Attack, and Release settings don't matter.
- With bi-amping, the Lo X Freq parameter (20 Hz to 1 kHz) sets the crossover frequency between the low and high bands. In fig. 5.4, the crossover frequency is 500 Hz.
- The Hi X Freq parameter (1 kHz to 20 kHz) is 20 kHz. So, there are two bands—below 500 Hz, and above 500 Hz (i.e., the frequency range between the Lo X Freq and Hi X Freq).
- To filter out the lows, the high band's Mid and Hi Gains are 0.0 dB. The Lo Gain is -60.0 dB.
- To filter out the highs, the low band's Lo Gain is 0.0 dB. The Mid and Hi Gains are -60.0 dB.

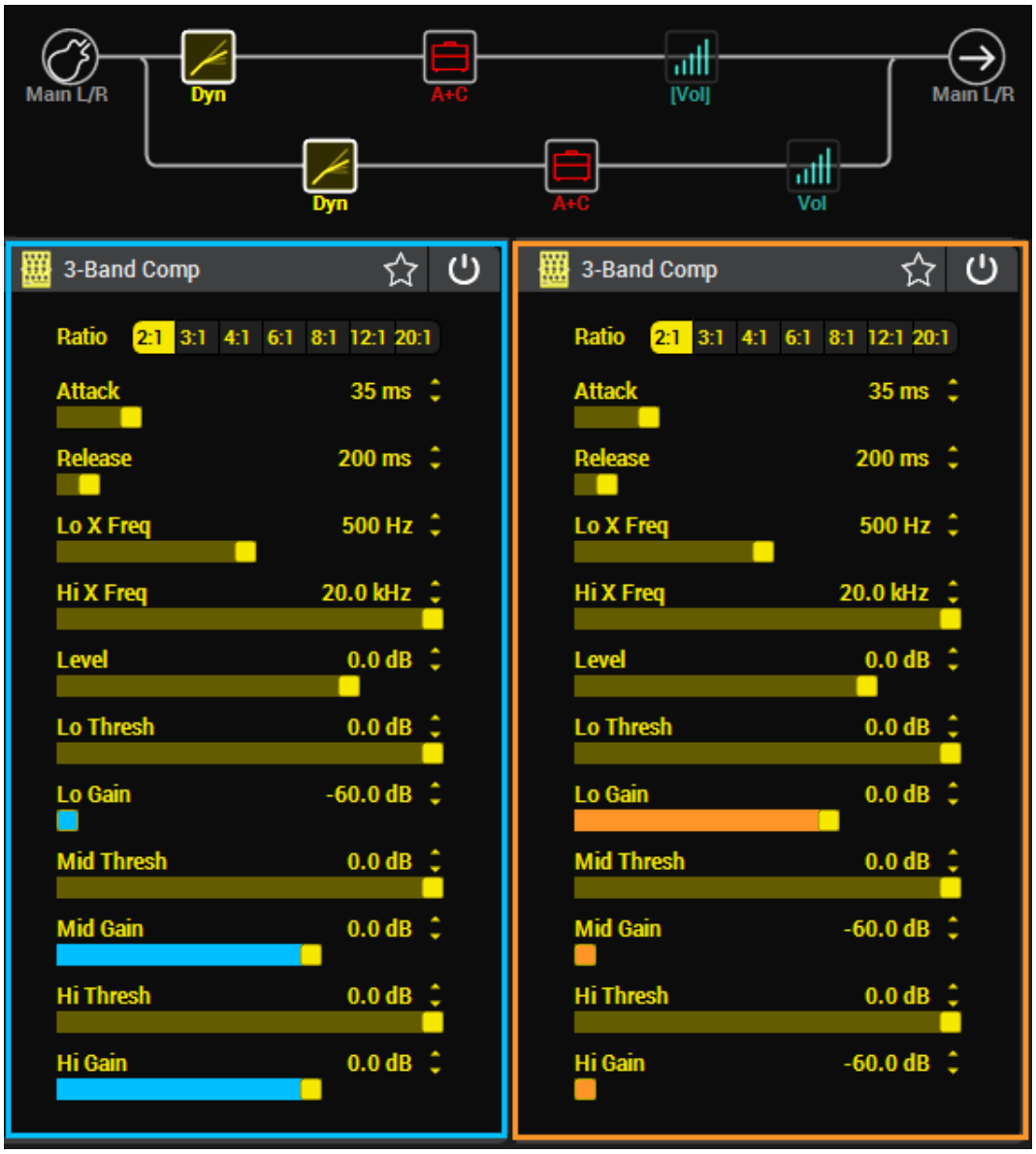


Figure 5.4 The 3-Band Comp splits the incoming audio across two independent paths: the high band (blue outline), and low band (orange outline). The parameters are identical, except for those indicated by the colored sliders.

Hybrid Bi-Amping + Multiband Processing

For many applications, this is my favorite bi-amping method for amps. It doesn't use the traditional bi-amping technique of splitting into high and low frequencies. Instead, it combines bi-amping (because it requires two amps) and multiband techniques. The multiband aspect involves sending five frequency bands into one amp, and a different set of five frequency bands into the other amp. This method works with all Helix models.

Fig. 5.5 shows the basic layout:

- Place the same amp in parallel.
- Precede each amp with a 10 Band Graphic EQ
- Send *alternate* bands from the EQs into their respective amps.

Figure 5.5 Two 10 Band Graphic EQs create multiband sends to two amps.

Sending frequencies that are further apart into each amp inhibits the frequencies from interfering with each other. This reduces intermodulation distortion. Note that the EQ's output levels are up +6 dB, to compensate for less signal going into the amp.

One of the advantages of this method is stereo/mono compatibility. For the studio, pan the output Mixer's parallel paths oppositely to create a stereo image. For the stage, because there's no delay or phase-based processing, summing to mono produces no notable change in sound quality. (The 10 Band Graphic EQ has some minor phase shifts because it models analog EQ, but you won't notice these in practice.) The 10 Band Graphic EQ settings create a flat response when summed to mono.

The amount of sonic improvement from this technique depends on what you play. For example, you probably won't notice much difference playing single-note solos with clean amps. But you'll hear a major difference with big, full chords going through distortion that include 7ths, 9ths, etc. Basically, the more frequencies that come out of the guitar, the more you'll hear a significant improvement.

Bi-Amping with Independent Paths

Because 4-Path Helix devices have two independent paths, one can be dedicated to the low frequencies, and the other to the high frequencies. This allows for more options:

- There's more processing power per path than if the bands were sub-paths within Path 1 or 2.
- Paths 1 and 2 can each have two independent, parallel bi-amp sounds.
- Path 1 can have a parallel bi-amp sound, while Path 2 has a single sound with lots of effects.

- Create a bi-amp setup in Path 1, and route Path 1 and 2 in series. Then there's enough DSP for lots of effects before and after the bi-amped sound.

Input Split Block with 4-Path Helix

The Input Split block crossover can't split across Paths 1 and 2. It can split only across Paths 1A and 2A, or 2A and 2B. To use the Input Split crossover in a 4-Path Helix, insert an Input Splitter in Paths 1 and 2. Turn on for one of them, and turn down the A Level in both Mixers (fig. 5.6).

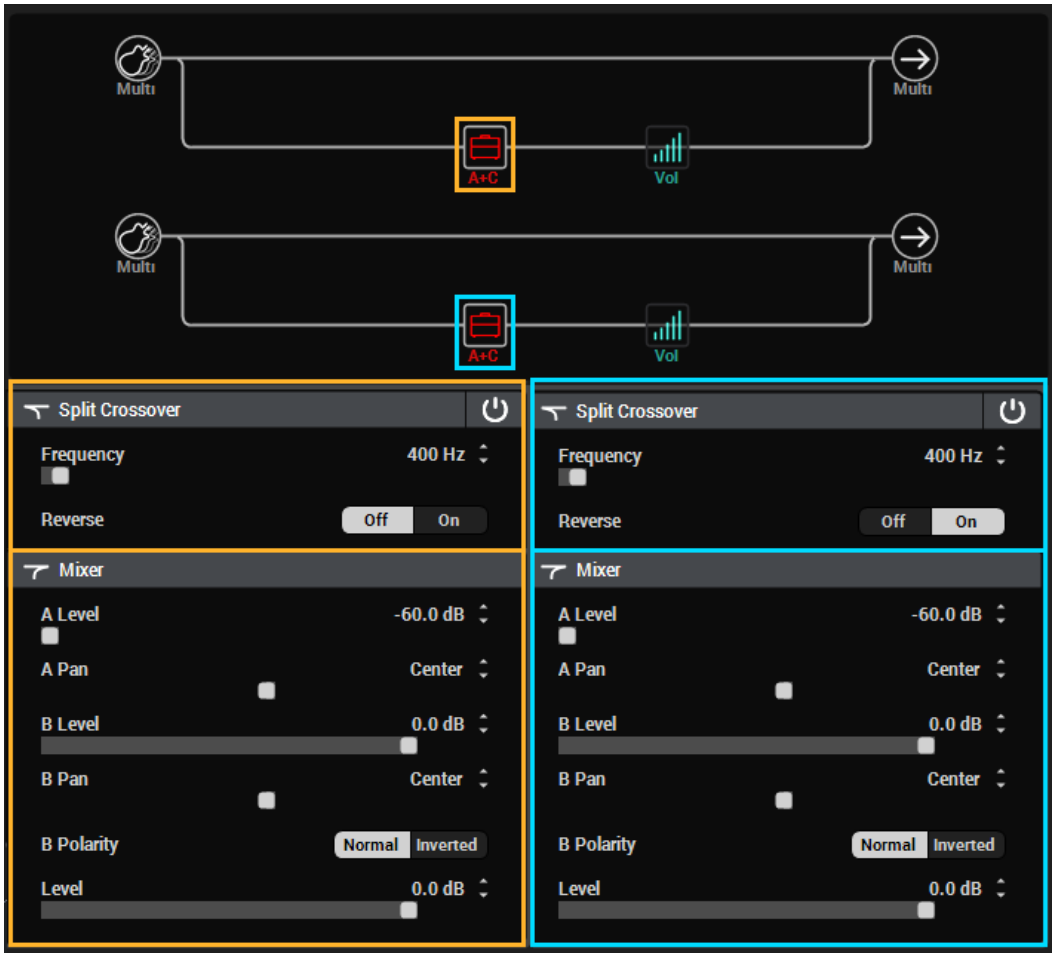


Figure 5.6 Input Split Crossovers provide one Split for each Path. Only one sub-path is used with each Path.

In the template **2B.TemSplitXover**, both splits are set to the same frequency (400 Hz as a default). The Path 1 Split (orange outline) sends the low-frequency band to Path 1B, which has an amp and mute button. At Path 1's Mixer, the A Level is turned all the way down. So, only the low frequencies are present at the Mixer output.

The Path 2 Split (blue outline) enables the Reverse parameter. So, the high frequencies go to Path 2B, which has an amp and mute button. At Path 2B's Mixer, the A level is turned all the way down. Only the high frequencies are present at the Mixer output.

Other 4-Path Bi-Amp Setups

For the Shelf Split, 3-Band Comp Bi-Amp Split, and Hybrid Bi-Amping+Multiband Processing routings, use Path 1A and Path 1B of a 2-Path Helix as Paths 1 and 2 of a 4-Path Helix, respectively.

Share a Cab

Bi-amping two Amps+Cabs uses more DSP power than a single Amp+Cab. One way to free up processing power is to mix two Amps together, and then send the Mixer output to a Single Cab. Sharing a cab sounds almost as good as individual cabs. Single cabs are mono. So, they can't preserve any stereo imaging from panning in the Mixer.

To have different cabs for the two amps, insert a Dual Cab after the Mixer. Pan the Mixer inputs hard left and right, so each amp goes to its own cab. Dual Cabs require more DSP power than a Single cab. The section Converting Single-Band Presets to Dual-Band Presets shows one way to take advantage of sharing a cab.

Tip: I'm not sure this tip is all that useful. But y'all are creative, so you might find some cool application. The concept is to split into two paths with Amp+Cabs, but not split into high and low bands going into the amps. Instead, split into high and low bands at 500 Hz *after* the amps. For the high band, set the associated cab's Low Cut to 500 Hz, and High Cut to Off. For the low band, set the associated cab's Low Cut to Off, and the High Cut to 500 Hz. This creates a super-wide stereo spread that might be useful for…something. The same concept works with a Dual Cab inserted after the Mixer.

Bi-Amp Templates

The *Free Files > Chapter 5 Presets* folder includes eight bi-amp templates:

- Templates with an *underscore* after the number of bands (e.g., **2B_TemSplitXover**) are compatible with HX Stomp and other two-path devices.
- Templates with a *period* after the number of bands (e.g., **2B.TemSplitXover**) are compatible with 4-path Helix family devices.
- All templates incorporate Volume/Gain Mute Button blocks in the two paths. When creating a preset, the mute buttons can isolate a band. This makes it easy to edit one band without being distracted by the other band's sound. After optimizing the two bands, you can delete the Vol blocks.
- The templates include Amp+Cabs so you can hear something immediately after loading the template. The Amp+Cabs were selected arbitrarily. They're not optimized or recommended as the "best" cabs.

2B.TemSplitXover is a 4-Path template that uses the setup in fig. 5.6.

To use the following 2-Path templates with 4-Path devices, copy paths 1A and 1B to Paths 1 and 2, respectively.

2B_Tem3BandDyn uses two 3-Band Compressors to split the audio into high and low bands (see fig. 5.4). The compressor crossovers are set to 500 Hz. This template uses the most DSP power, but an advantage is that you can compress a frequency band by lowering the band's Threshold and adjusting Ratio, Attack, and Decay.

2B_TemHybrdBiAmp uses a medium amount of processing power. It gives the cleanest distortion sounds of all the templates, because it reduces intermodulation distortion the most effectively. This template may not be as useful with 4-Path setups, because the goal behind this 2-Path process is to replace a single Amp+Cab with a cleaner, single version.

2B_TemShelfBiAmp also draws a medium amount of processing power. It provides a subtle differentiation between high and low frequency bands. It's well-suited to effects.

2B_TemSplitXover uses the Input Split crossover function to provide the high and low bands. The crossover frequency is 450 Hz. This template uses the least DSP power, because it doesn't require adding any blocks.

Hearing Is Believing

Bi-amping produces a different kind of sound. The goal isn't to duplicate an existing amp's sound (you already have an amp for that, right?), but to create something new. Usually, the sound is less "rough," and more articulated, than a single Amp+Cab. One feature of bi-amped distortion is that if you hit a big chord, notes played high up on the neck at the same time sound more distinct.

The following examples work with all Helix family members (except HX Effects).

Bi-Amping with Preamps vs. Amps

The Brit Plexi Jump is one of the Helix's most DSP-hungry amps. But with some workarounds and the more efficient version 3.50 cabs, we can still do bi-amping and have DSP left over for effects. These presets exemplify bi-amping with power chords, which are a challenging audio source for amp sims to handle. Our goal is more focused, articulated variations on the standard Amp+Cab.

1. As a reference, load the preset **1B_Brit PlexiJmp**. This inserts the standard Brit Plexi Jump Amp+Cab between the input and output.
2. Play some power chords for a minute or so. Note how they sound.
3. Next, load the preset **2B_Brit PlexiJmp** (fig. 5.7). This preset bi-amps paralleled Brit Plex Jump Amps, with a 600 Hz crossover frequency. Version 3.50's more efficient cabs allow both paths to have an Amp+Cab. To free up DSP power for additional effects, use amps, and then mix to a single cab after the Mixer.

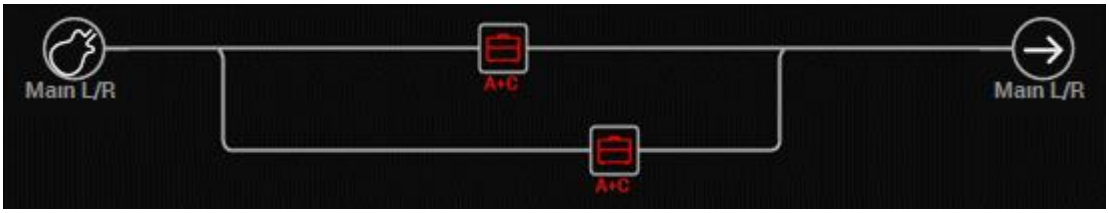


Figure 5.7 Bi-amping the Brit Plexi Jump Amp produces a smoother sound compared to a single amp.

4. Start playing, and you'll hear that the sound is more focused than a single Amp+Cab. This isn't to say it's "better"—it's more like two facets of the same amp. Both sub-paths are panned to center for a fair comparison with the **1B_Brit PlexiJmp** version.
5. Experiment with inserting a wah. Experiment with placing the wah before or after either amp in either path, before the Input Split, and after the Mixer. The results are quite different for these six different placements.

Next, load the preset **2B_BrtPlxJmpPrFX** (fig. 5.8). Unlike the previous preset, the crossover is set to 1.1 kHz so that it's more "lead-friendly." For a beefier tone try 650 Hz. Overall, the tone is less rough, with more separation between the high and low frequencies. Again, this doesn't mean it's "better." It means that between these two Brit Plexi Jump presets, we've obtained different, usable sounds that still retain the amp's essence.

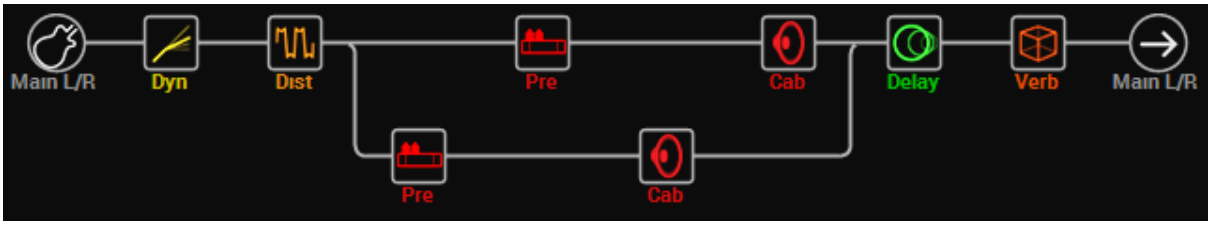


Figure 5.8 By using preamps instead of amps, even with bi-amping the Brit Plexi Jump blocks it's possible to add several more effects *and* have stereo imaging and reverb.

2B_PVitriolCrnch shows off another Preamp application using the Input Split Crossover. The PV Vitriol Crunch requires so much DSP that although you can have two Amps in parallel, you can't add anything else other than a Volume/Pan block. Even sharing a Single cab isn't possible. Of course, you don't *need* to do bi-amping—a single Amp+Cab sounds pretty cool. But bi-amping gives alternate sounds you can't get otherwise.

So, insert two PV Vitriol Crunch Preamps in parallel, each with its own cab. The Preamps don't sound all that different from the equivalent Amps. This configuration allows for plenty of unused DSP power (fig. 5.9). As with other presets that use the Input Split Crossover, experiment with the input crossover frequency for the desired sound.

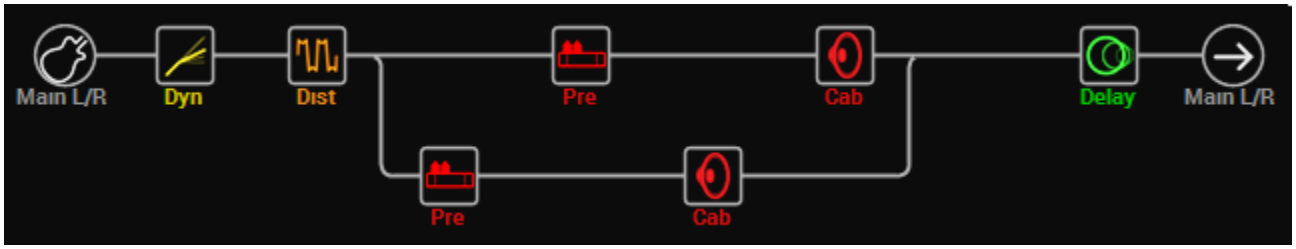


Figure 5.9 Even with a bi-amp PV Vitriol Crunch sound, there's still enough DSP power for a stereo Dual Delay at the output, your choice of several distortions (the preset uses the KWB), and a Deluxe Comp at the chain's beginning.

This preset is ideal for stereo. The Mixer pans the Cabs in stereo, and the Delay is stereo. However, this preset sounds great in mono too. There's tight gain-staging, so enabling/bypassing any of the three effects maintains the same perceived output level.

The previous two presets show that whether preamps sound like amps or not, they certainly come close. Adding a cab and additional effects often obscures any significant differences.

Bi-Amping with Distortion

Splitting a distortion effect into two separate bands prior to feeding two parallel amps can improve the overall clarity and smoothness.

1. For reference, load the preset **1B_Grammatico** (fig. 5.10). This uses the Grammatico Nrm Amp and 1x12 Grammatico Cab. It's preceded by the Top Secret OD distortion.

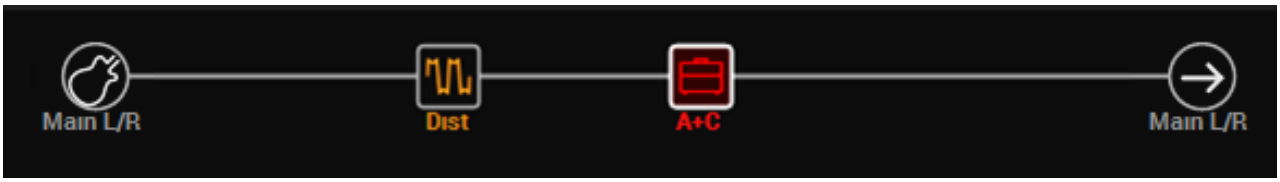


Figure 5.10 This preset provides a basis of comparison with the bi-amped version.

2. Enable and bypass the distortion to get a feel for the sound.
3. Now load the bi-amped **2B_Grammatico** preset (fig. 5.11).
4. With distortion bypassed, the sound isn't too different from the 1-Band preset, aside from the bi-amping. But with distortion enabled, the sound is more distinct and defined. This is because the distortion isn't shoving all its frequencies into one amp.
5. To become familiar with how changing the Split Crossover's Frequency parameter affects the sound, click on the Input Split to open it, click on Split Crossover, then vary the frequency between about 200 Hz and 3 kHz. The sound changes considerably at different frequencies.

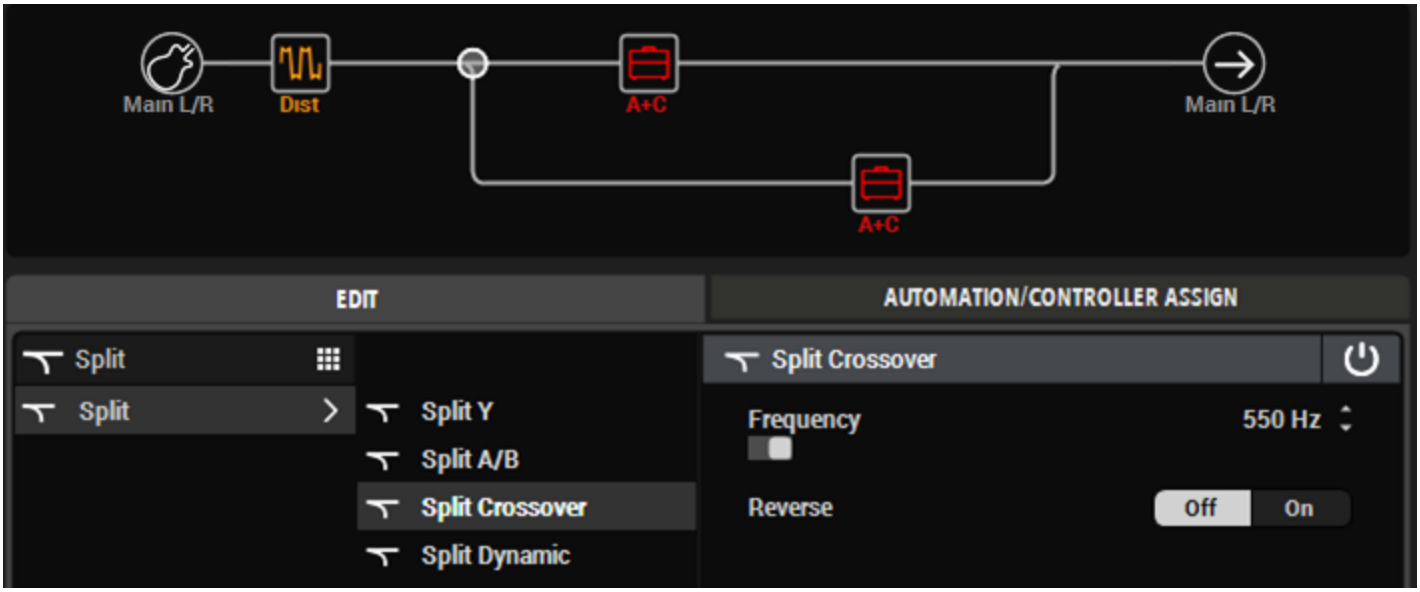


Figure 5.11 Bi-amped Grammatico, preceded by Distortion.

Bi-Amping the Essex A15 for Greater Articulation

This is one of my favorite amps. Bi-amping delivers a more focused sound and allows processing the high and low frequencies separately.

1. Load the preset **1B_Essex A15**. This uses the default A15 Amp parameters and adds a reverb.
2. Play some power chords. Note the sound quality.
3. Now, load the preset **2B_Essex A15** (fig. 5.12). This splits the audio into high- and low-frequency paths. Each goes into its own Essex A15 block.

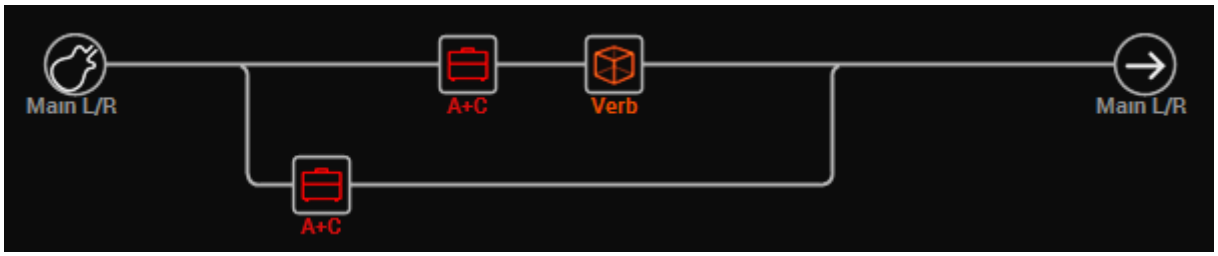


Figure 5.12 Splitting into two bi-amped blocks allows adding reverb on only the high-frequency path.

The sound is more articulated, and often fits into a busy mix better than a single Amp+Cab. However, you may need to turn up the level a bit for the same impact.

Only the high-frequency split has reverb. The ambiance doesn't affect the lower frequencies, which can potentially muddy the sound. Although Helix reverbs often have tone controls or low-cut/high-cut filters, the resulting sound isn't quite the same.

Bi-Amping with Different Amps

Bi-amping also works with clean and mildly crunchy amps, not just distorted sounds. Load the preset **2B_WhoUSDoublNrm** (fig. 5.13).

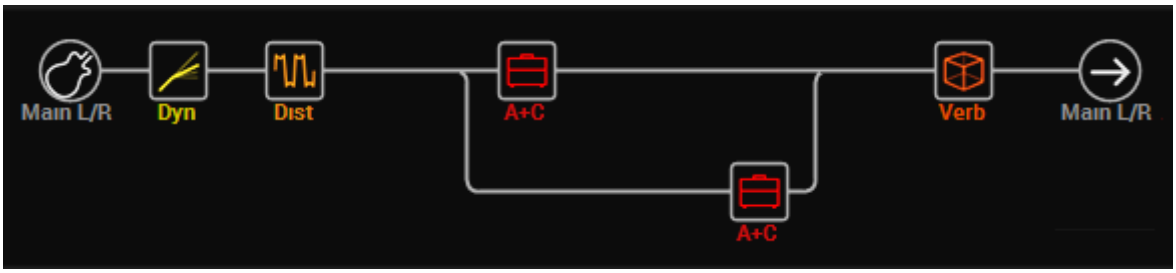


Figure 5.13 Bi-amping can be effective when combining slightly overdriven sounds and relatively clean amps.

The Deluxe Comp at the input evens out the dynamics. The Teemah! distortion gives a mild overdrive. The higher-frequency band uses a WhoWatt 100 (another one of my favorite Helix amps) followed by a 2x12 Double C12N Cab. The low-frequency band has a US Double Nrm Amp, followed by a 4x12 Greenback 25 Cab to make the low end a little heavier. Adding Dynamic Ambience at the end gives a solid room sound.

Editing Tips

One bi-amping advantage is being able to optimize the amp and cab parameters for their specific frequency ranges. For example, sometimes isolating the highs emphasizes "fizziness." Lowering the high-frequency cab's High Cut can reduce this. To concentrate on editing a path, remember that you can insert a Volume/Gain block Mute Button to mute the other path.

We've only scratched bi-amping's surface:

- Use different mics on two cabs.
- Choose different Amp+Cab blocks for the two paths.
- After the Mixer output, insert the Mono2StDualDela or Mono2StSmplDela in the *Free Files > Chapter 11 Favorites* folder. These use the Dual Delay or Simple Delay blocks respectively to provide a wider stereo image.
- Process only the high- or low-frequency band.
- Add one of the simpler EQs at the output for general tone shaping.

Although 3- and 4-band processing takes multiband processing further (see Chapter 7 and Chapter 9), bi-amping will often do what you need.

Converting Single-Band Presets to Dual-Band Presets

After hearing my multiband Helix preset pack, some people asked whether they could convert their existing presets to multiband operation. The answer is…maybe. The main challenge is available processing power. Multiple amps and cabs—the heart of the multiband process—consume more processing power than effects. Although the efficiency of the version 3.50 cabs helps lower DSP consumption, there may not be enough remaining DSP power to retain DSP-hungry effects like the Cosmos Echo.

This underscores the strengths and limitations of the Helix's dynamic effects allocation. Although a preset might run out of processing power, making some minor tradeoffs might reclaim enough power to create the sound you want. The following example shows the kind of tradeoffs that can convert a single-band preset into a bi-amped one.

Fig. 5.14 (top image) shows an older preset with a Tone Sovereign distortion, Guitar+Legacy A30 Fawn Brt Amp and 2x12 Bluebell Cab, stereo 4-Voice Chorus, and stereo Cosmos Echo. If you try to insert a second

Legacy A30 Fawn Brt Amp+Cab in a parallel path for bi-amping, it's not possible. You have to change the effects to ones that require less DSP.

Fig. 5.14 (bottom image) shows the same basic preset, modified for bi-amp operation—and more—thanks to the new 3.50 cabs. After going through the Input Split crossover for a bi-amped split, the two amps mix into a Dual Cab with two 2x12 Bluebell cabs. The amps can pan in stereo before the Mixer, and the Dual Cab preserves that stereo imaging. Furthermore, each cab has its own mic and mic placement for a big, stacked sound.

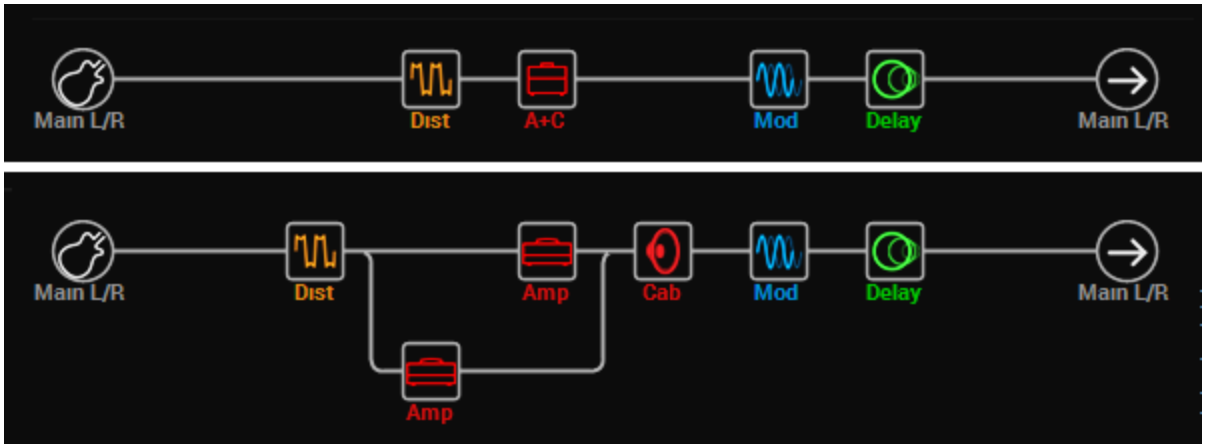


Figure 5.14 The top image shows a single-band preset based on the Legacy A30 Fawn Brt. The lower image modifies the preset with the version 3.50 cabs for bi-amp operation.

Chapter 6 | CABINET TECHNIQUES

Cabs are an important part of amp sounds, but they also provide creative options. Far from being “set and forget,” cabs can contribute much to innovative sounds.

Cab Options

Clicking on a cab in the Edit pane presents four options (fig. 6.1):

- **Single.** Versions 3.50 and 3.60 introduced 33 cabs. We'll call them **New** cabs, as opposed to pre-version 3.50 **Legacy** cabs. New cabs have improved sound quality yet require 60 to 70% less processing power. Note that Single cabs are mono. Stereo audio at the cab input will be mono audio at the output.
- **Dual.** This loads two New cabs in parallel. Unlike Legacy cabs, they can be panned in the stereo field. Mono or stereo audio feeding a Dual Cab will be dual mono or stereo audio at the output, respectively.
- **Single Legacy.** These are pre-version 3.50 Helix cabs. They remain available to retain compatibility with pre-version 3.50 presets.
- **Dual Legacy.** These also use pre-version 3.50 Helix cabs. Functionally, they work similarly to the new Dual Cabs. However, panning is hard left/hard right. More flexible panning requires an extra block.

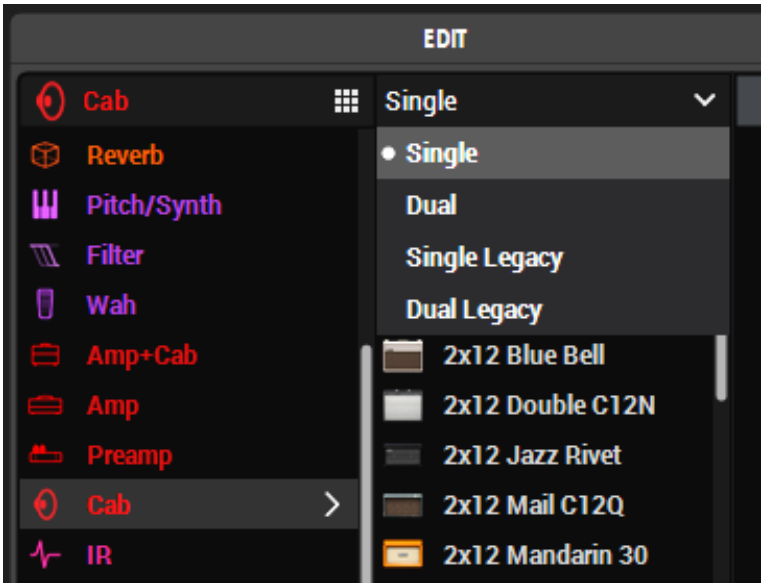


Figure 6.1 Helix now has four primary cab categories.

Linking/Unlinking Dual Cabs

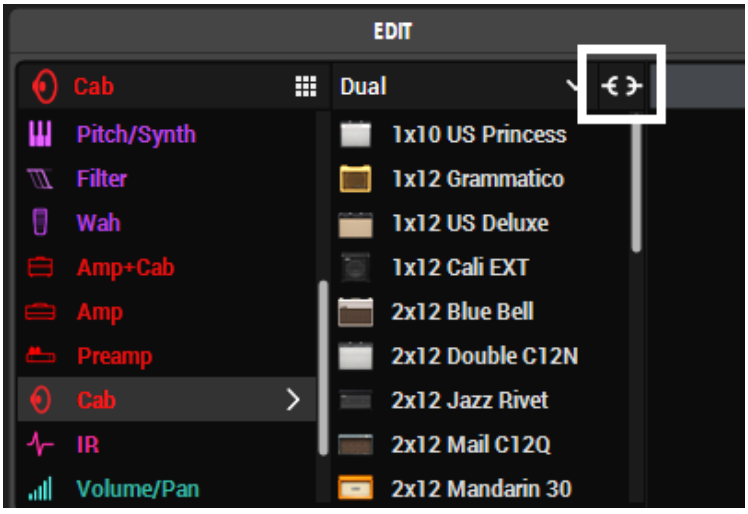


Figure 6.2 The Link/Unlink button is outlined in white.

When first selecting a Dual Cab in the Edit pane (Legacy or New), you can choose to Link or Unlink the selection (fig. 6.2). Linking loads two of the same cabs. This is useful for combining different mics or mic placements with the same cab sound. However, after loading you can choose a different cab for either cab. Click on the tab with the cab's name. Choose a cab from the Edit pane.

Dual Cabs cannot combine Legacy and New cabs. You can add only the same cab type loaded originally.

Unlinking the selection loads the cab you selected as the first cab. The second cab defaults to the 2x12 Jazz Rivet for New Dual Cabs, and to the 1x12 Lead 80 for Legacy Dual Cabs. After loading, you can choose a different cab for either cab, as mentioned above.

Amp+Cab Options

Clicking on an Amp+Cab in the Edit pane presents four options (fig. 6.3):

- **Guitar** pairs the selected guitar amp with a New cab. If there's no revised version of the original cab, Helix selects the 3.50 cab that gives the best match.
- **Bass** works like Guitar, but with the version 3.50 bass cabs.
- **Guitar+Legacy** pairs the selected amp with the default Legacy cab.
- **Bass+Legacy** pairs the selected amp with the default Legacy cab.

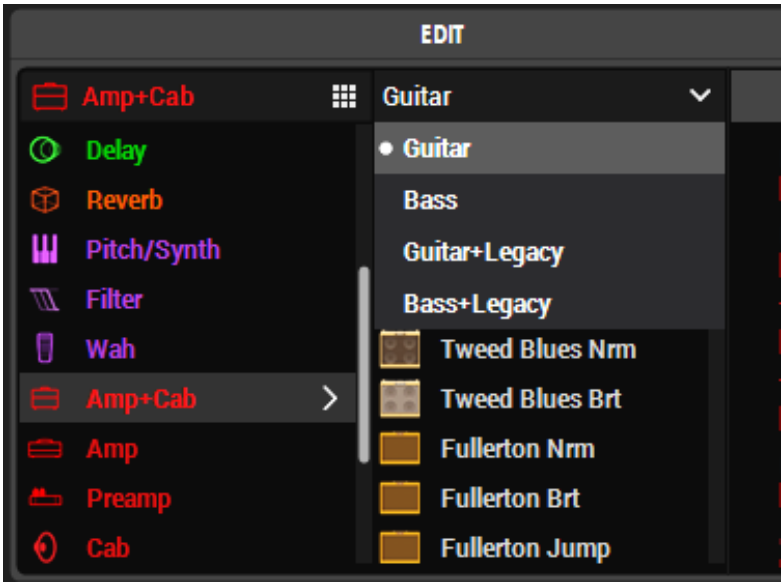


Figure 6.3 Amp+Cab options introduced in version 3.50.

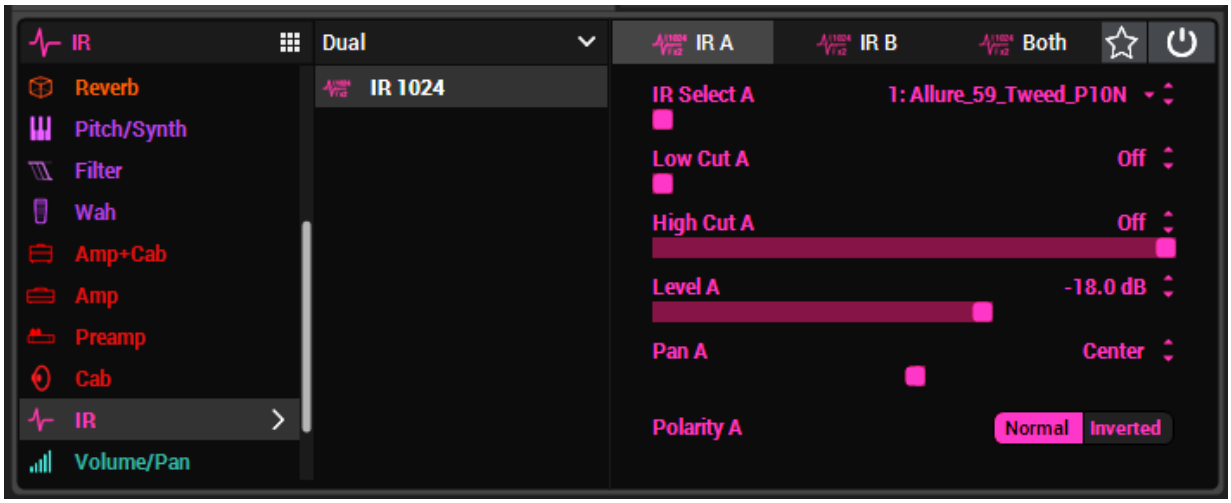
IR-Based Cabinets

IRs (Impulse Responses) "sample" a cab's sound with one or more mics and mic positions, and turns the result into waveforms that can be loaded into blocks. They've become somewhat of a cottage industry for Helix aficionados. Although users debate stock cabs vs. IRs, either one can provide tones the other can't. Some users favor using only stock cabs for compatibility, while others prefer IRs. Later in this chapter, we'll show how to create your own IRs from other amp sim cabinets.

If you're not familiar with the procedure for loading IRs, the owner's manual describes the process clearly. Hardware Helix units require using **HX Edit**. Helix Native has its own **IR Panel**.

Single and Dual IR Blocks

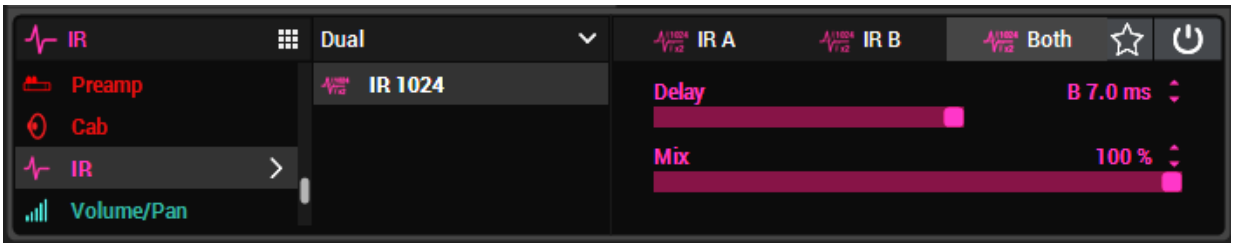
- **Single** IR blocks (named Mono IR blocks prior to v3.50) can import 1,024 or 2,048 point IRs.
- **Dual** IR Blocks can import two 1,024-point IRs (A and B impulses).



Click on the IR A tab to edit A impulse parameters, or on the IR B tab to edit B impulse parameters:

- The **IR Select** drop-down menu selects the IR file to load into the selected slot.
- **Low Cut** reduces low frequencies for the selected slot. Choose a cutoff frequency from Off to 500 Hz.
- **High Cut** reduces high frequencies for the selected slot. The cutoff frequency covers 500 Hz to 20 kHz.
- **Level** sets the output level for the selected slot's IR.
- **Pan** positions the selected slot's cab output in the stereo field. With Helix floor, pressing the knob resets pan to Center. With Helix Native, Ctrl/⌘+click resets to Center.
- **Polarity** flips the polarity of the IR's waveform. If enabling an IR block thins the sound, the IR could be out of phase compared to anything else that's in parallel (either the other IR in a Dual IR block, or a processor in a parallel signal path). Inverting the polarity may produce a fuller sound.

The Both Tab



The Both tab's parameters affect both slots:

- **Delay** is like a fine-tunable version of the normal/invert button. Some IRs might be partially out of phase, instead of the 180 degree phase shift the normal/invert button fixes. Turning the hardware's knob counterclockwise from center (or moving the Helix Native slider to the left) delays the A slot IR by up to 50 ms. Going past center in the other direction delays the B slot IR by up to 50 ms. You usually adjust this setting for the fullest possible sound. However, thinner timbres may be musically useful.
- **Mix** controls the IR block's wet/dry mix. With cab IRs, the preferred setting is 100%. With non-cab impulses like acoustic guitar body resonances, you may want to blend dry and processed sounds.

Cab Frequency Responses (1.5)

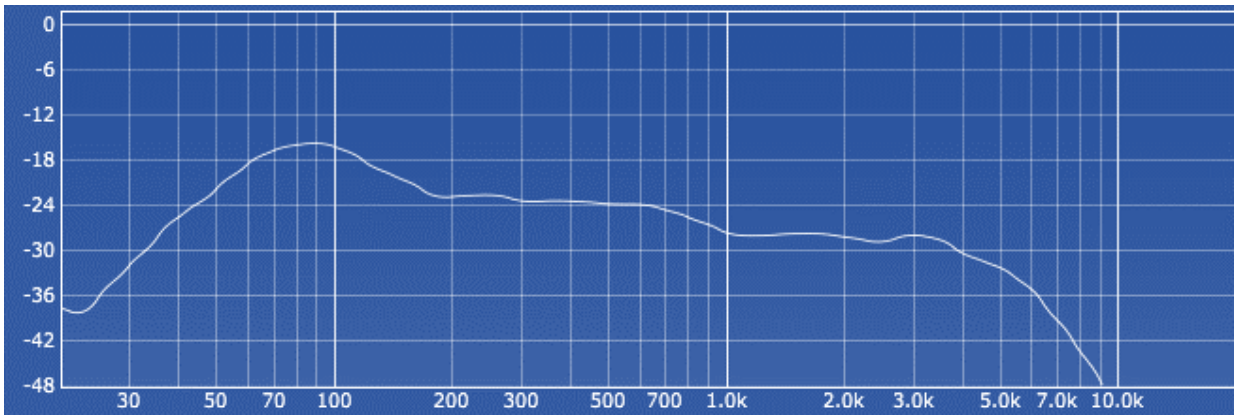
Version 3.50 introduced 20 guitar cabs and 4 bass cabs based on a better-sounding, more efficient cab engine. 3.60 and 3.70 each added 7 guitar cabs and 2 bass cabs. 3.80 added 4 guitar cabs. Cabs are listed in order of the number of speakers and speaker size. The Bass cabs are toward the end, starting with the 1x12 Epicenter cab.

The following frequency response curves highlight the sonic characters of these 46 cabs. Your ears will decide which cab to use, but these curves can speed up the process. For example, if you want a scooped cab that emphasizes highs, the curves can guide you to potential candidates.

To standardize the comparison, all **Guitar Cab** curves use the same setup: 121 Ribbon mic, Center position, 2-inch Distance, and 0° angle. **Bass Cabs** use the 414 Condenser mic, with the same other parameters as the guitar cabs. Any Low Cut or High Cut settings that were part of the default preset remain unchanged.

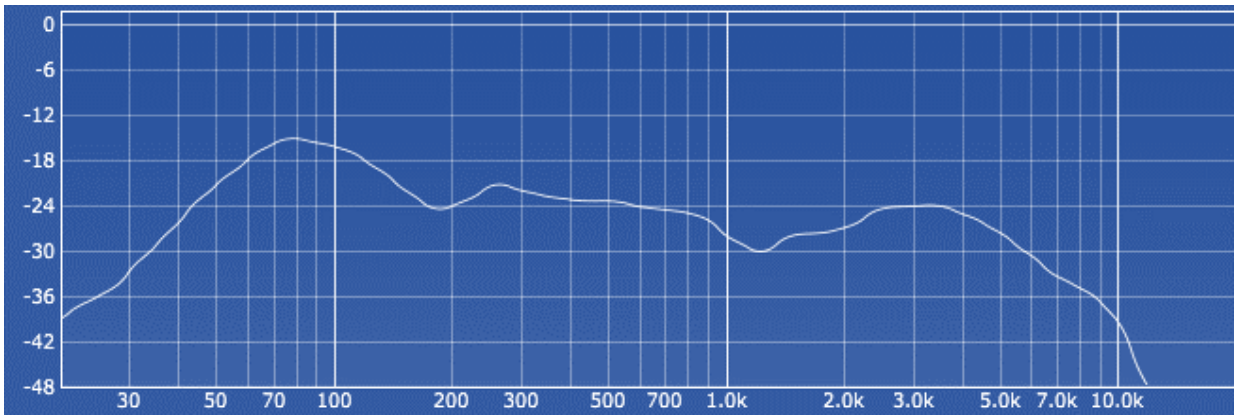
Disclaimer: There are also some subjective comments. However, remember that as soon as you make even slight changes to mics, amps, etc., a cab that I say I like for dry leads might be perfect for metal power chords.

Soup Pro Ellipse



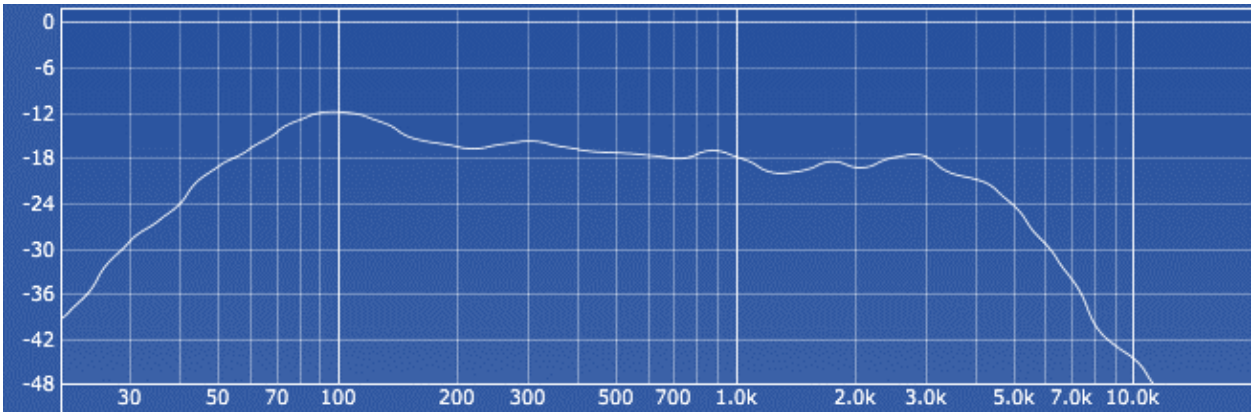
The basic character is a beefy, bass-heavy sound. To reduce the proximity effect and provide more balance over the frequency spectrum, the default mic setup moves a condenser mic further back and somewhat away from the cone.

1x8 Small Tweed



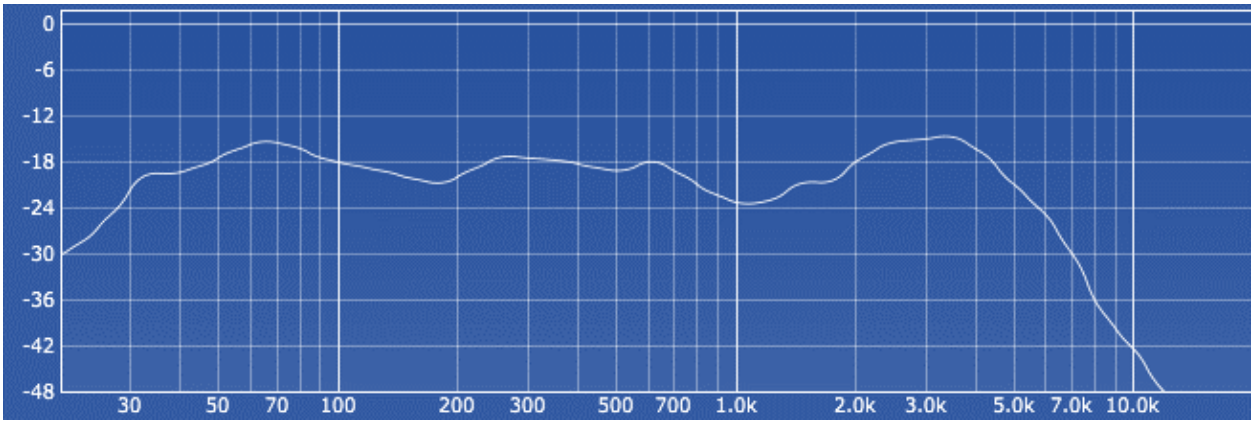
Along with the hefty bass response, the dip around 1 kHz reduces “honk,” while the 200 Hz dip reduces potential tendencies toward mud. The peak centered at 3 kHz gives more definition and brightness. The default miking moves the mic a bit further away to reduce the proximity effect.

1x10 US Princess



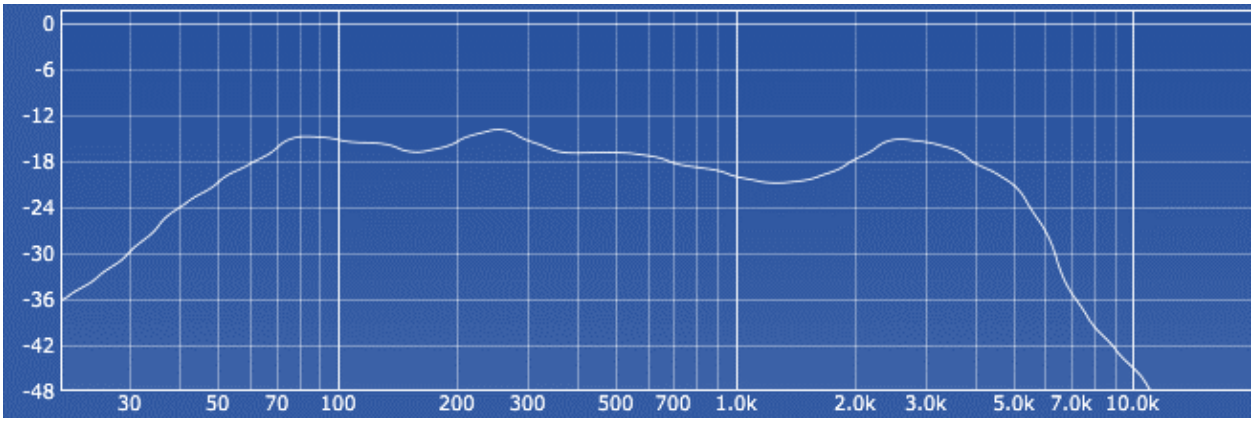
From 200 Hz to 3 kHz, the response is almost flat. The 100 Hz bass emphasis gives more warmth. Unlike cabs with a major upper midrange lift, this one (around 3 kHz) is mild. The cab works well for warm, defined rhythm guitar.

1x12 Fullerton



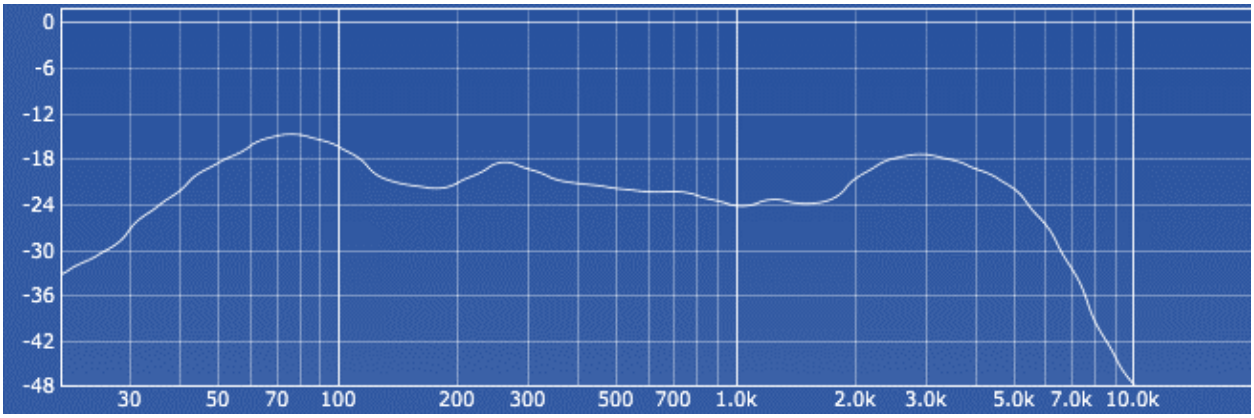
Try this cab for bright sounds with presence and articulation. The scoop around 1 kHz reduces “honk.” The bass extension is more pronounced than many cabs, but not overpowering.

1x12 Cali IV



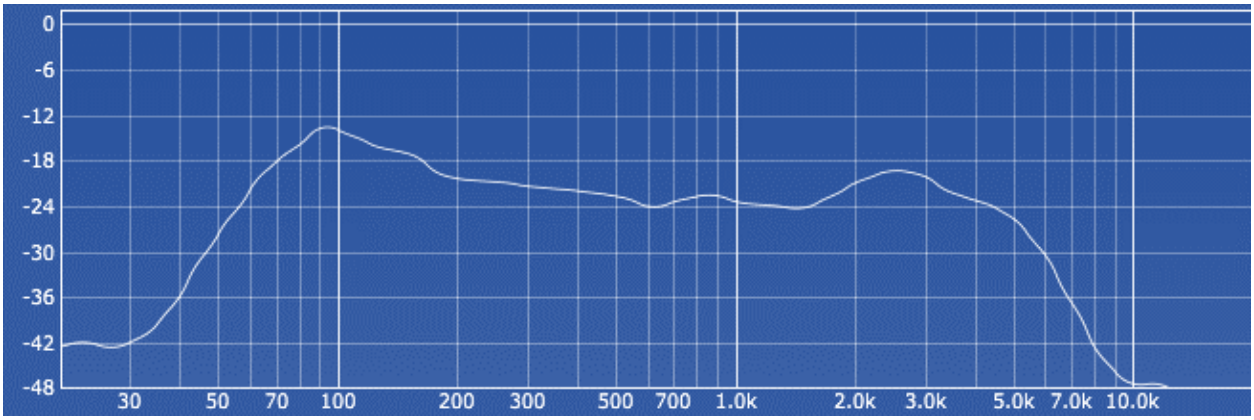
This has many of the same characteristics as the 1x12 Fullerton. However, the 1x12 Cali IV has a flatter response, less bass extension, and a slightly higher-frequency scoop.

1x12 Grammatico



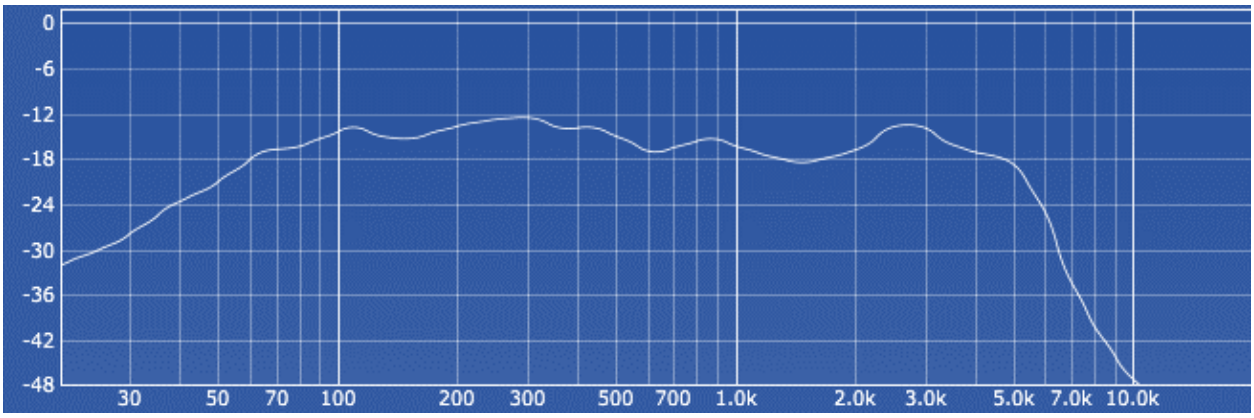
The dip around 200 Hz reduces “mud” and balances out the bass emphasis. From there, the response gradually declines up to about 2 kHz. The significant upper-midrange peak increases definition and helps cut through a live or studio mix. However, these frequencies can also be a little harsh. Any harshness may sound more pronounced with single-coil pickups than with humbuckers.

1x12 US Deluxe



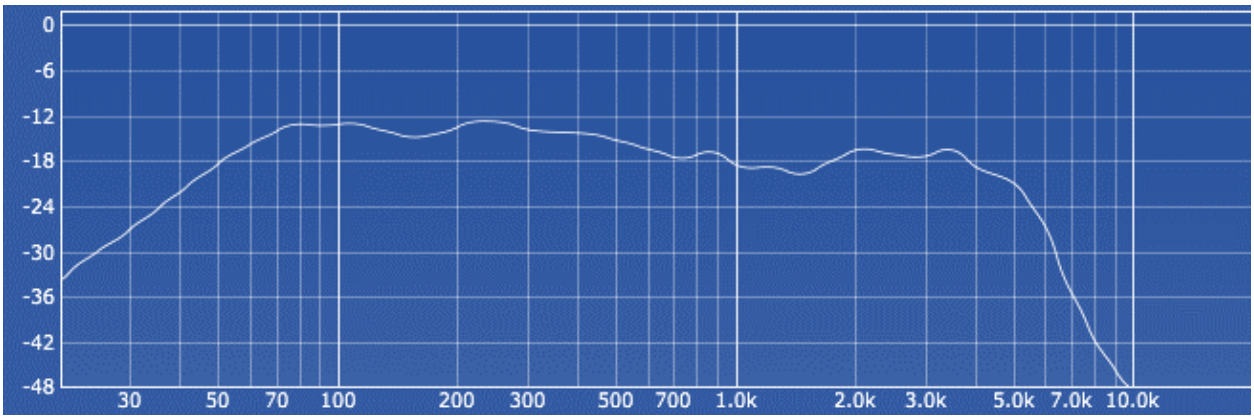
There’s emphasized bass, a shallow/broad scoop, and upper midrange peak around 2.5 kHz for a little extra definition. The peak is narrower than many other cabs. This helps bring out lead parts while keeping brightness under control.

1x12 Open Cast



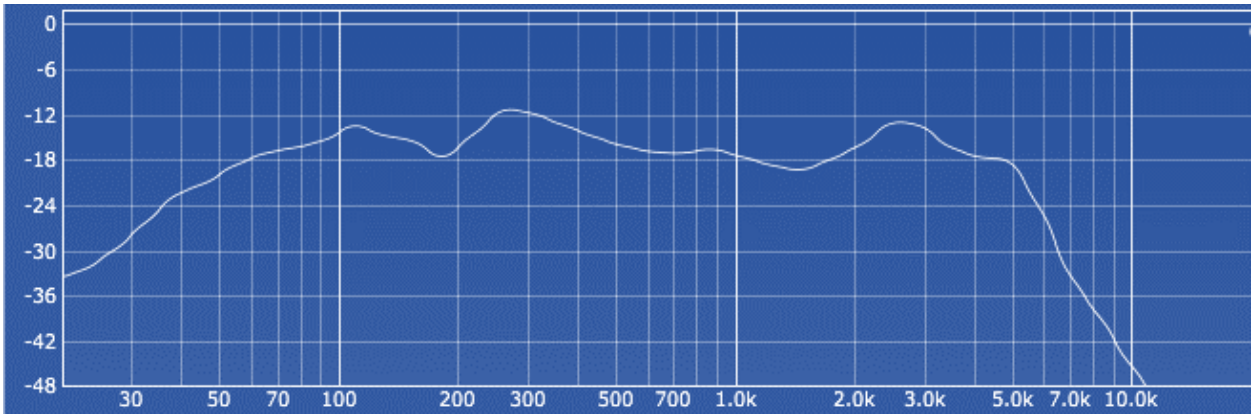
This custom open-back cab has a smooth, even response. The upper mid peak adds snarl and definition.

1x12 Open Cream



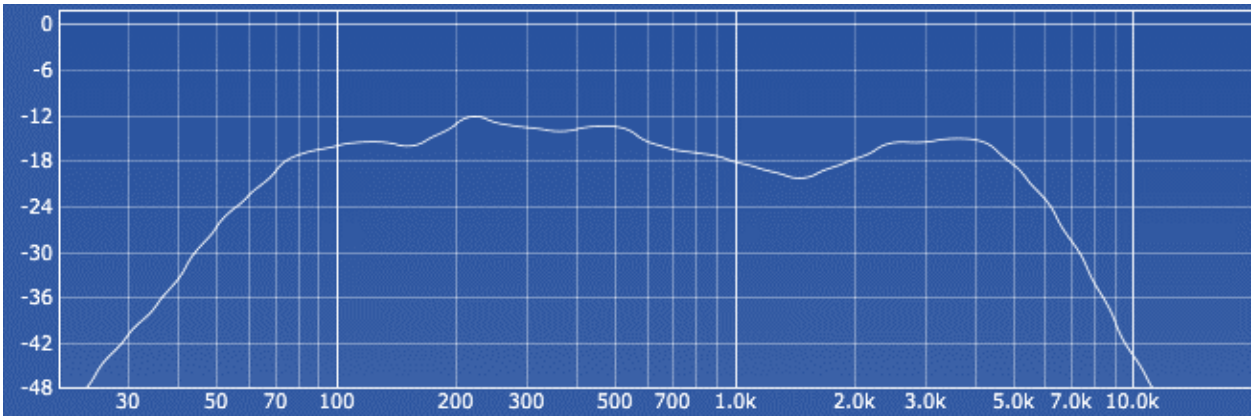
The response is like the Open Cast, but with less overall brightness and a more subdued high end. It's better suited to rhythm guitar parts than leads but may sound dull with dry humbuckers.

1x12 Cali Ext



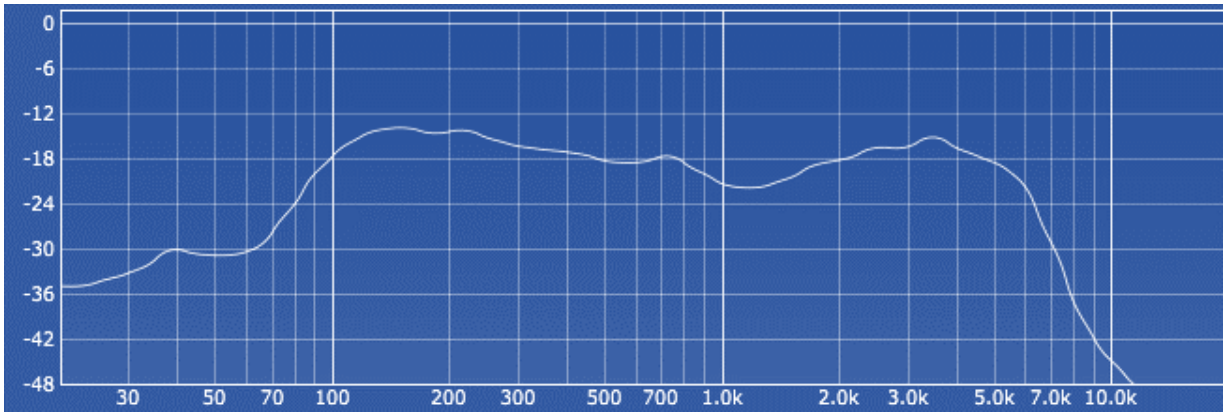
The upper midrange peak gives bright articulation, which contrasts with the shallow 1.5 kHz scoop. The twin response peaks at 100 Hz and 250 Hz contribute to a beefy lower midrange. The 180 Hz cut tames the lower midrange.

1x12 Blue Bell



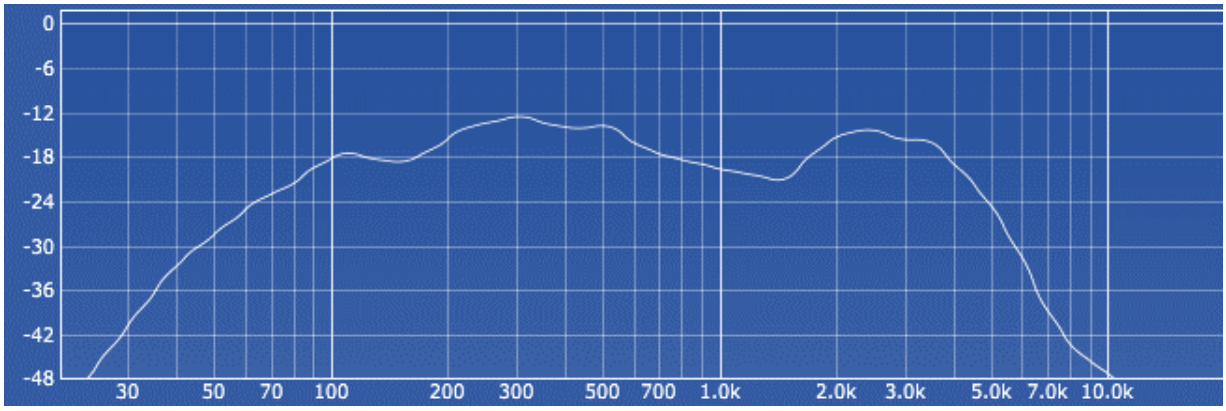
This gives a balanced "pop" sound. The low end is present but doesn't dominate, the dip around 1.5 kHz opens up space for vocals and other instruments, and the peak in the upper midrange enhances definition.

2x12 Interstate



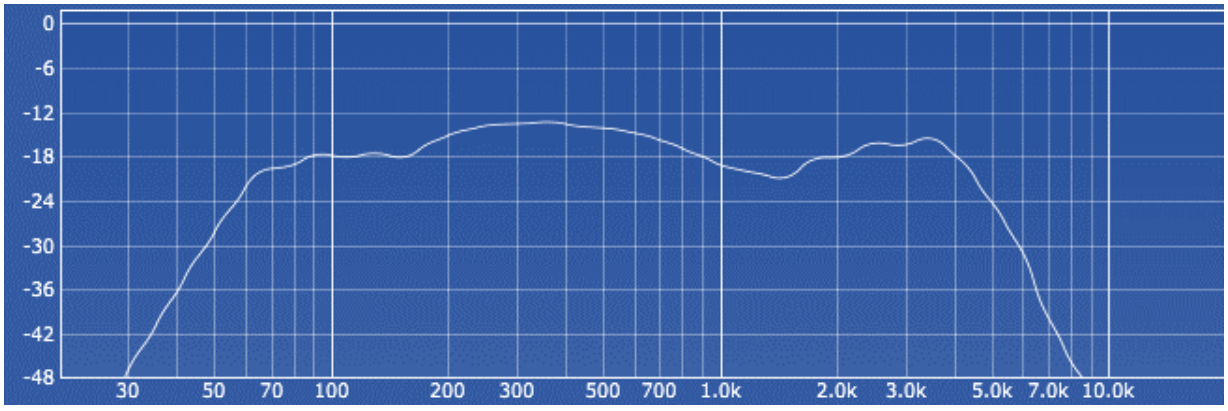
A low end lift around 100 Hz adds bass. Below the guitar's range, the bass drops off rapidly for a tighter sound. The midrange scoop and added presence complete this cab's focused character.

2x12 Blue Bell



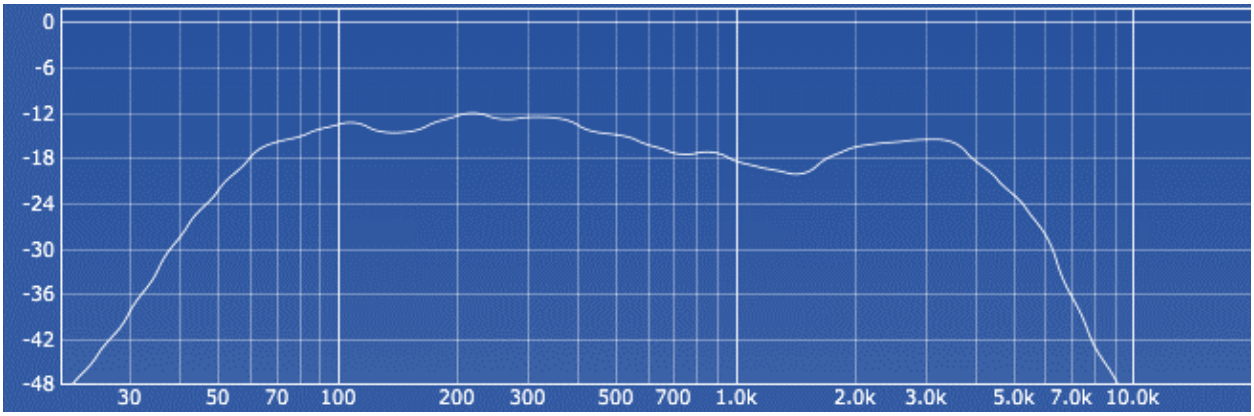
This character-laden cabinet features a major, broad peak in the lower midrange. The cut around 1.7 kHz de-emphasizes the "honk" frequency. Like the 1x12 Blue Bell, this makes room for vocals. The major peak between 2 and 4 kHz adds articulation, while the steeper high-frequency rolloff reduces fizz. It's a fine choice for when you want rhythm guitar to power a track, yet have enough definition for when you switch to lead.

2x12 Silver Bell



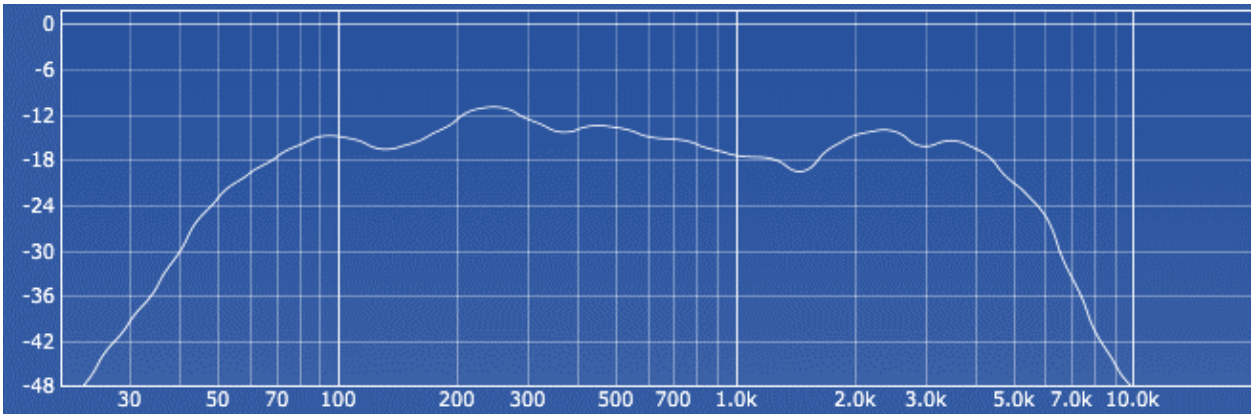
Compared to the 2x12 Blue Bell's response, this has a smooth, full overall response. It also trims the high- and low-frequency responses a bit more. It doesn't cede quite as much response to the vocal range.

2x12 Match G25



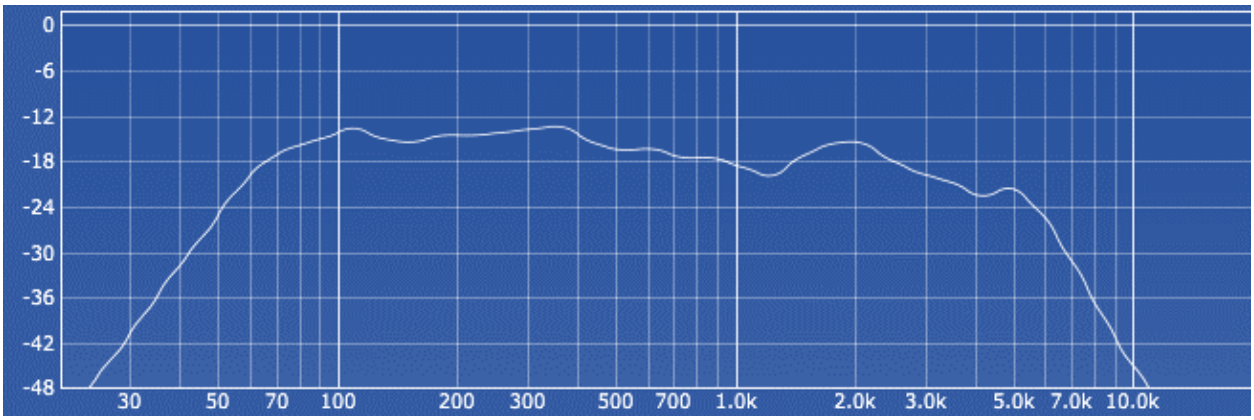
The rich lower midrange gives a full low end. The upper mid peak adds some definition. The rich, full sound is ideal for big rhythm guitar parts.

2x12 Match H30



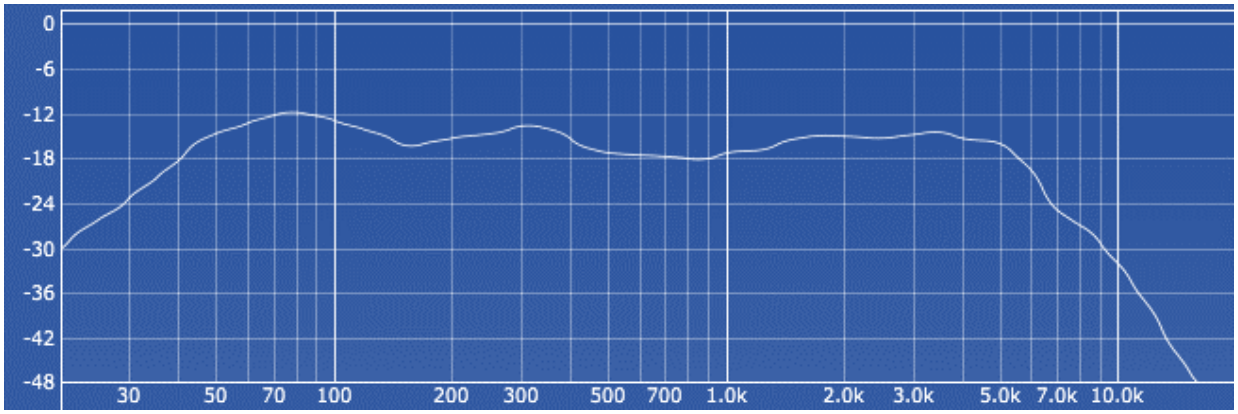
People often react to cabinets with significant frequency response variations as having more “character.” This goes through a roller coaster in the bass and lower midrange. At around 400 Hz, the response slopes downward to 1.5 kHz, only to climb again to a modest upper midrange peak. The high-frequency response extends further than many cabs.

2x12 Double C12N



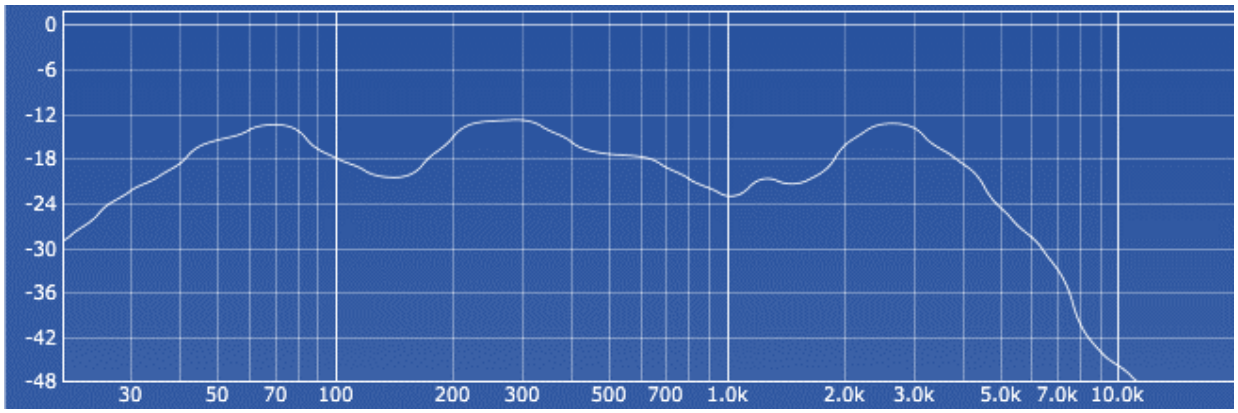
The dips at 1.5 kHz and 4 kHz don’t have too much impact on the sound, and the diminishing response between 2 and 4 kHz reduces potential harshness. Use this for a midrange-oriented sound that’s good for leads and rhythm.

2x12 Jazz Rivet



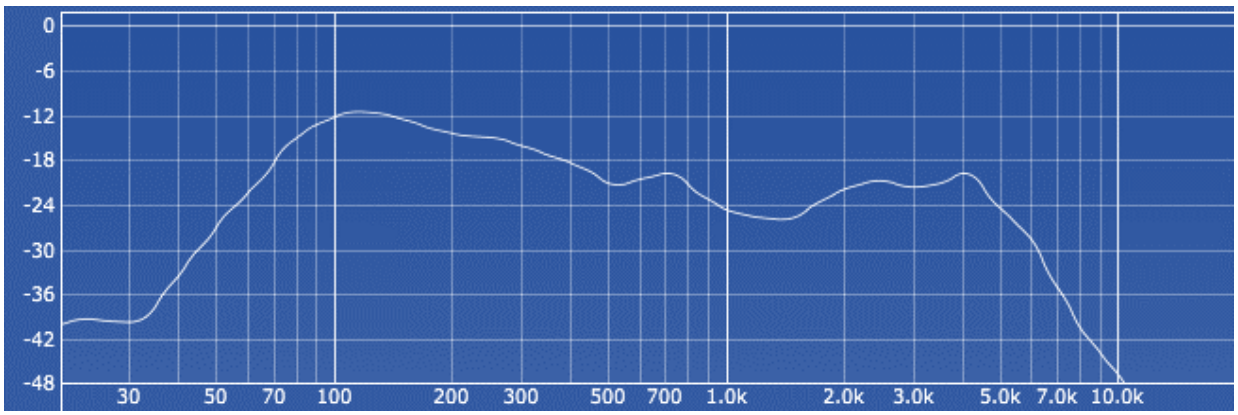
Note the deep bass response favored by jazz guitarists, and the relatively flat response. Use this cab for dry jazz guitar. The extended high end maintains definition, even when rolling back the tone control.

2x12 Mail C12Q



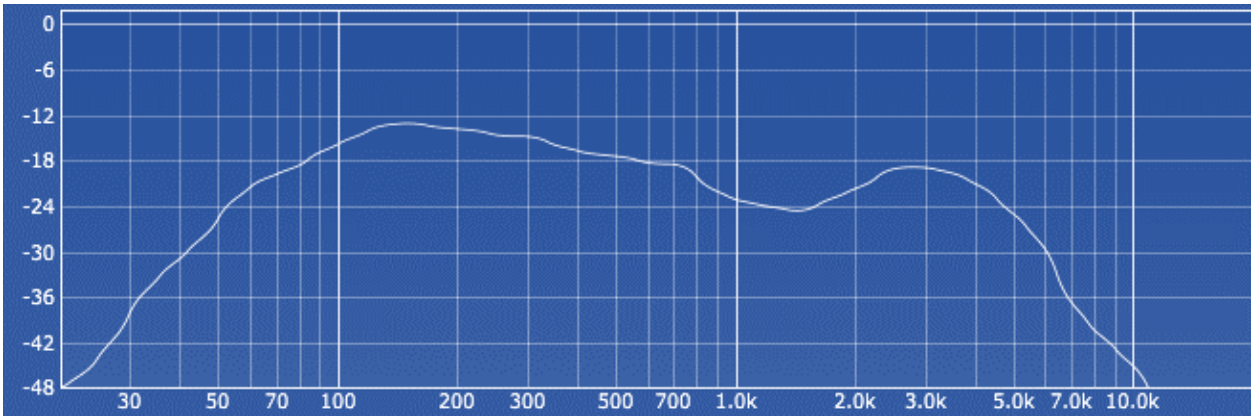
There are major peaks around 70 and 250 Hz, and in the 2 to 4 kHz range. With a fairly severe cut at 150 Hz, and scooped response from 500 Hz to 2 kHz, this cab has a well-articulated high end, space in the mids for vocals, lower-midrange beef, and enough of a 150 Hz dip to avoid “muddiness.” This is a useful response with octave dividers. The strong bass emphasizes the octave-below audio. The 150 Hz dip keeps it from dominating.

4x12 Greenback 20



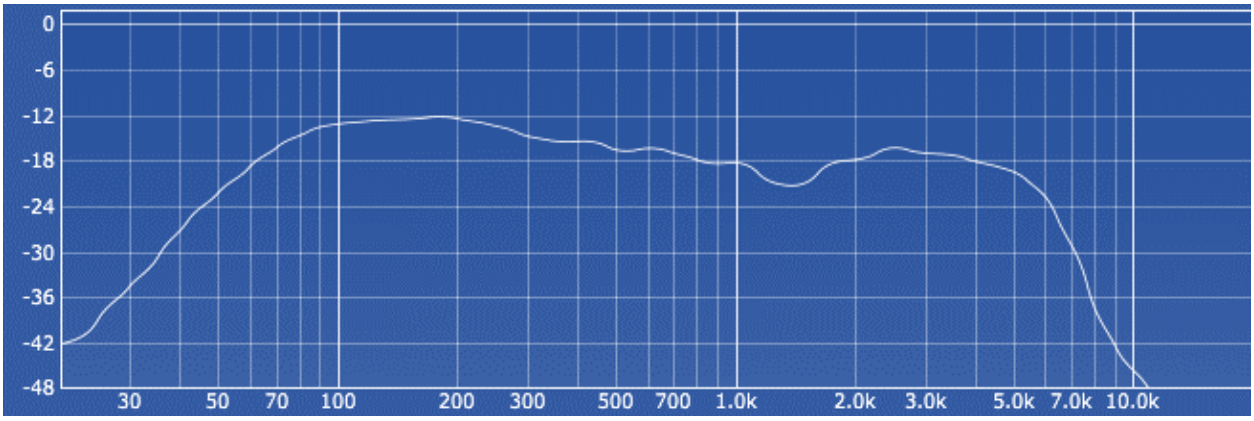
Marshall cabs have a specific character: major bass peak, rapid slope down to the 1.5 kHz range, and a peak in the upper mids that gives articulation while avoiding harshness. The full sound and decent definition works for lead or rhythm guitar.

4x12 Greenback 25



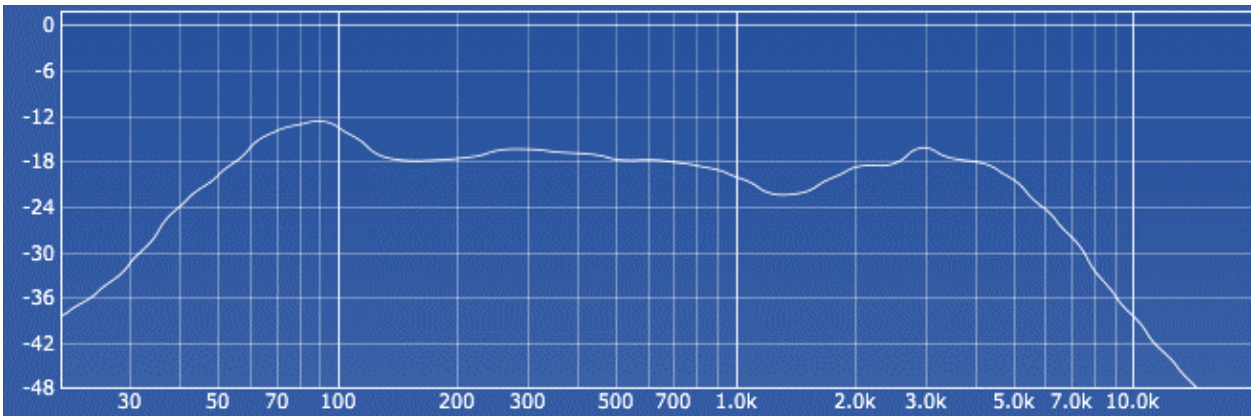
This resembles the Greenback 20, but has a smoother, more rounded response.

2x12 Mandarin 30



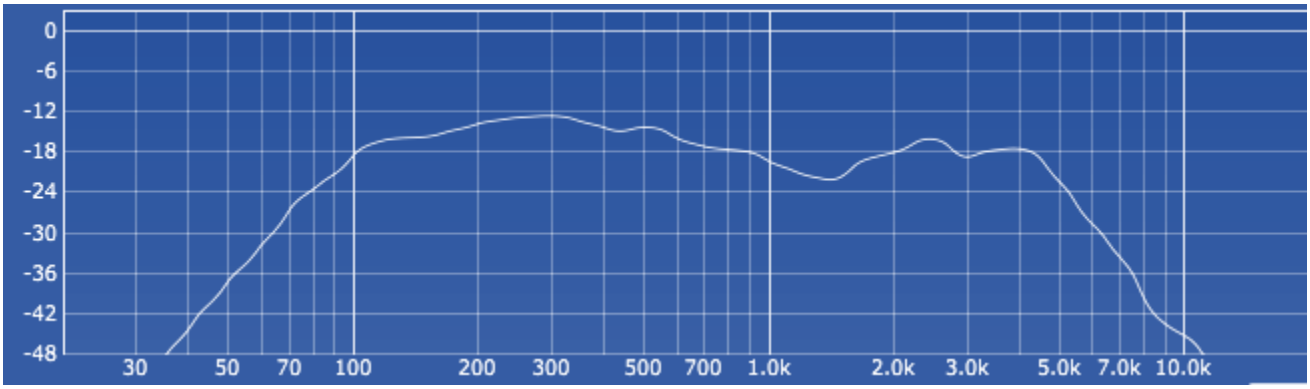
I like this one for leads. The emphasis in the upper-midrange articulation range is broad enough to avoid harshness. It might seem counter-intuitive that the prominent bass response works for leads, but remember that notes cover a wide frequency range. The full, big bass response can thicken solos that are higher up on the neck. With its full sound and defined presence, this cab is also a good choice for power chords and low tunings.

4x10 Tweed P10R



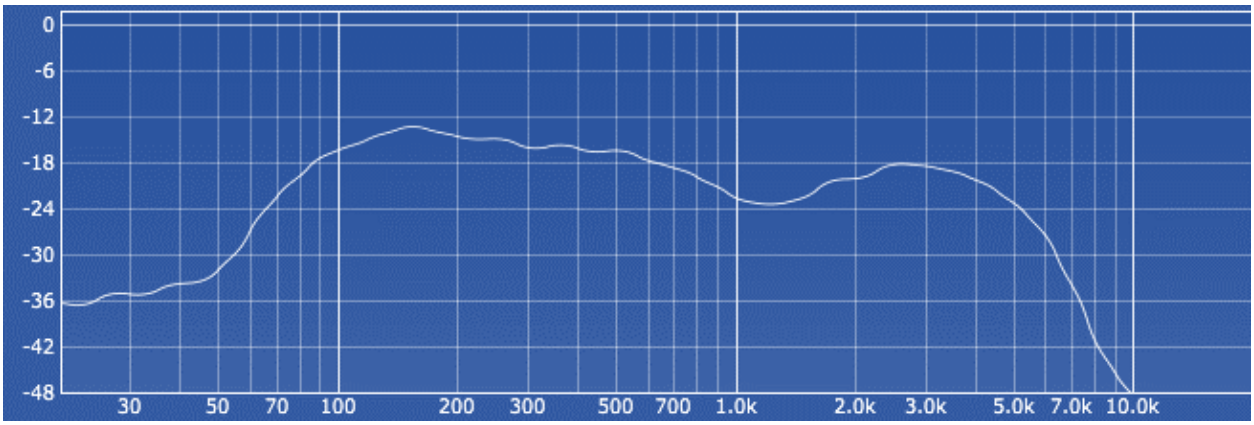
The uneven frequency response, with mild peaks and dips, adds some variety and character compared to a cab with a flatter response. Try this for clean, country-style leads (e.g., Tele).

4x10 US Super (1.5)



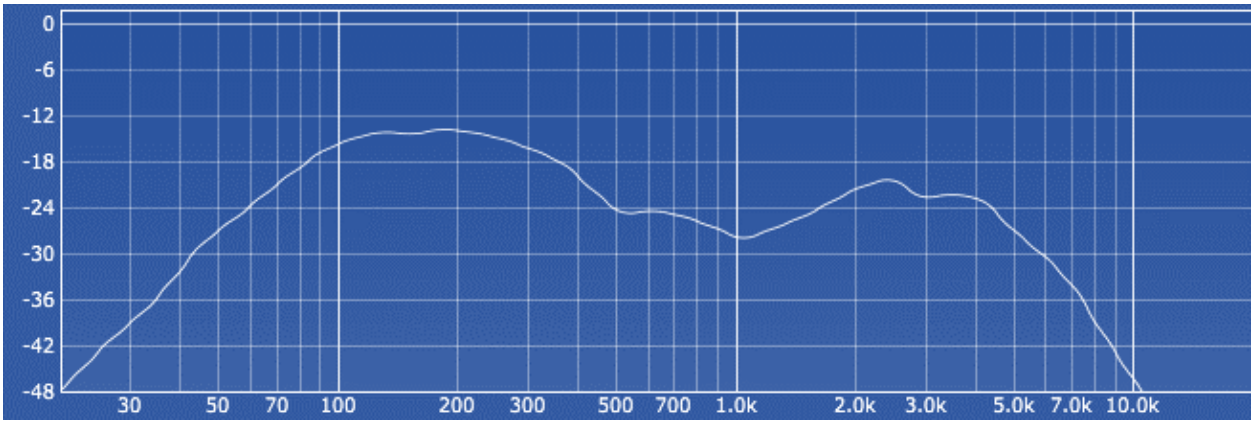
Compared to the 4x10 Tweed P10R, this de-emphasizes the low frequencies (the Cab also defaults to a 74 Hz Low Cut filter), has more midrange, and the high-end slope is a little steeper. There's still the slight scoop around 1.5 kHz.

4x12 WhoWatt 100



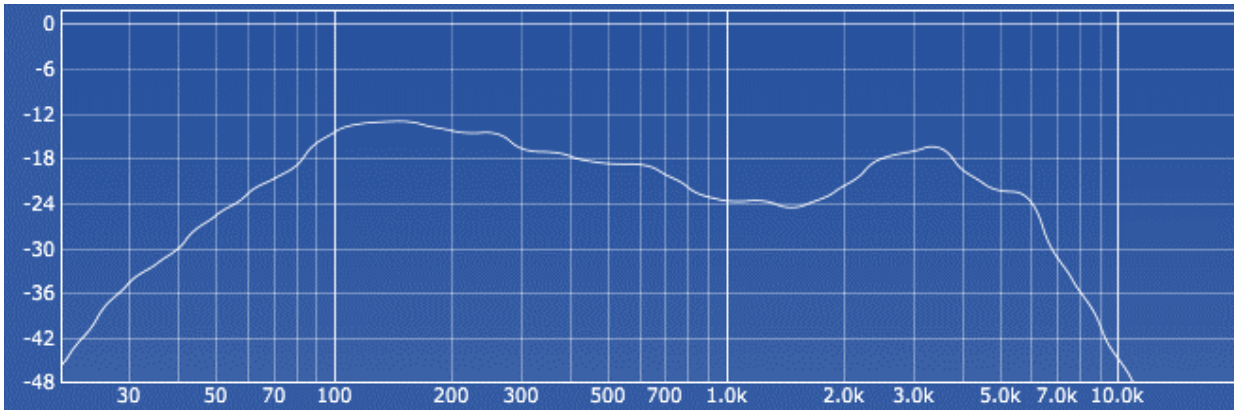
This isn't too different from the 2x12 Interstate, although the scoop is more pronounced compared to the high- and low-frequency lifts, and the overall response tilts toward bass.

4x12 Greenback 30



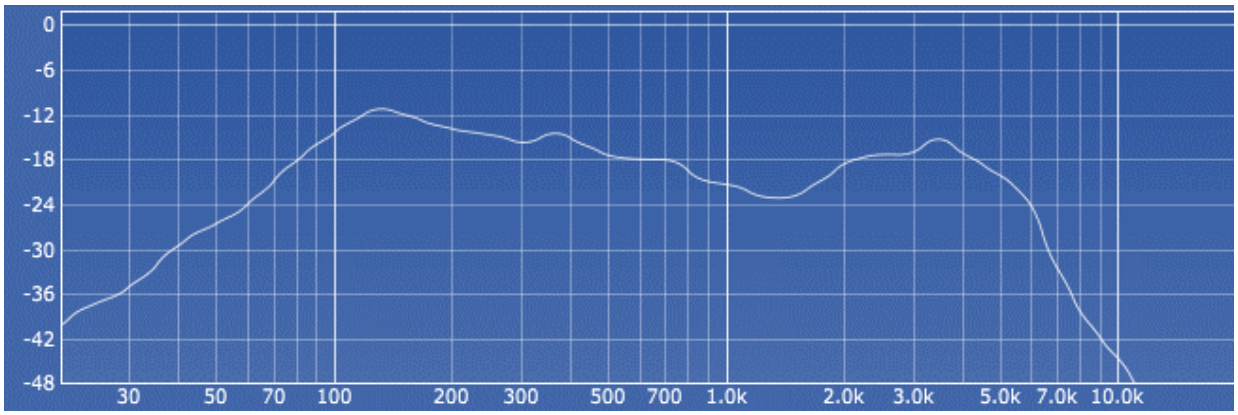
The low end delivers plenty of bass and beef, with a major midrange scoop. The moderate high-frequency emphasis counteracts the strong bass range.

4x12 1960A T75



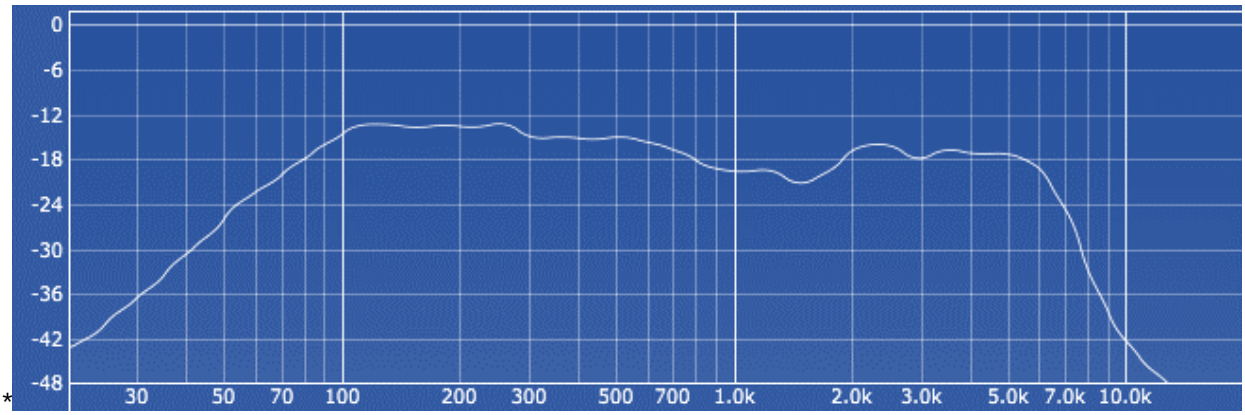
The scoop is a little broader than the previous amps. The uncharacteristic peak around 6 kHz adds extra zing for distorted leads, and sparkle for rhythm guitar.

4x12 Blackback 30



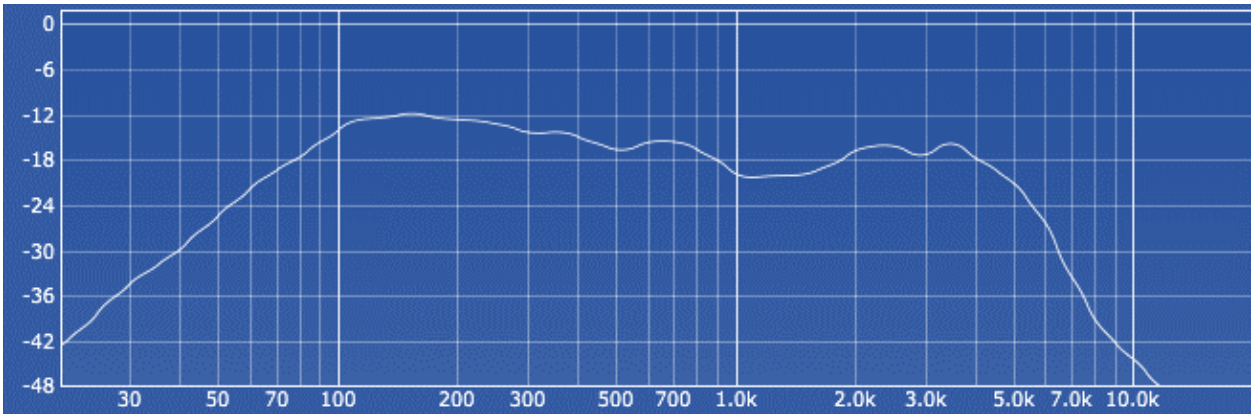
The similarity to the previous Marshall amps may be due to Park designer Mitch Colby being a veteran of Marshall and Vox amps. The main difference between this and the Greenback 25 is a broader high-frequency peak, and compared to the 1960A T75, a more present and even high end. Aside from those differences, the strong low end and gradual response falloff toward the midrange are characteristic of Marshall cabs.

4x12 Brit V30



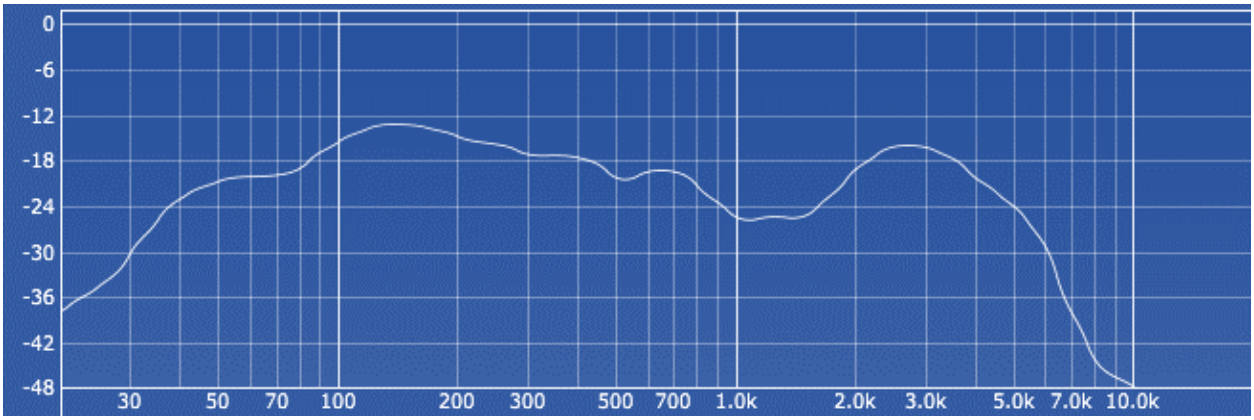
In this departure from traditional Marshall-world, there's an overall flatter response and less of a midrange cut. It's a good option for when you want a Marshall sound that dials back the bass a bit and has more brightness.

4x12 Cali V30



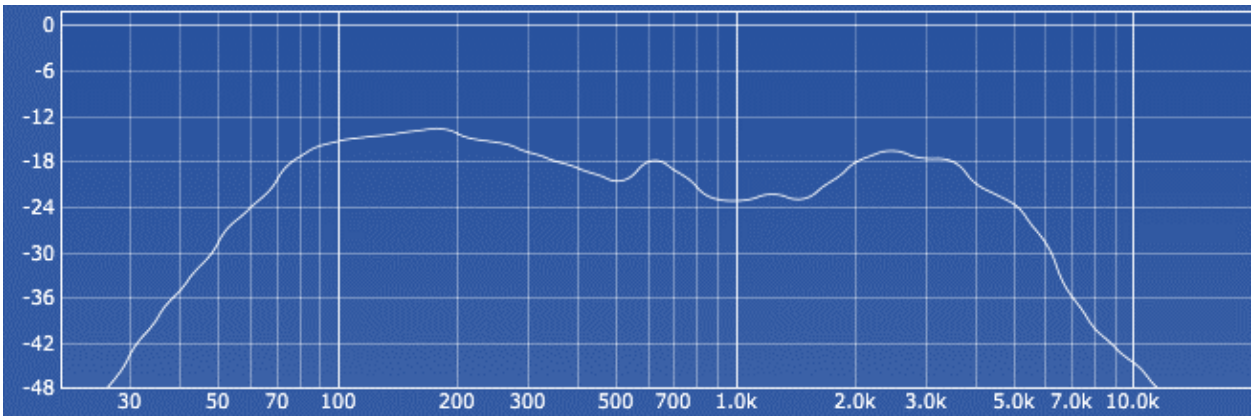
This cab resembles the Brit V30, but is a little more scooped, has more of a bump in the 2 to 4 kHz range, and the high frequencies drop off faster past 4 kHz. This is a smooth midrange performer. Note the slight bump around 700 Hz.

4x12 Mandarin EM



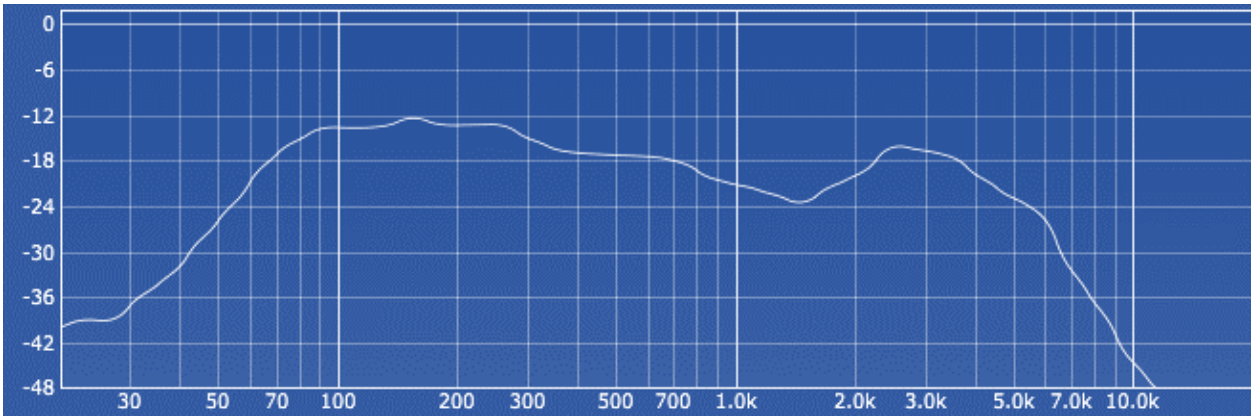
With a major scoop in the 1 to 2 kHz range, this is a twist on other 4x12 cabinets. The scoop's depth makes the high and low end more prominent. Also note the extended lower bass response. Leads can really sing on this, while leaving room for other instruments in the 500 Hz to 2 kHz range.

4x12 MOO)))N T75



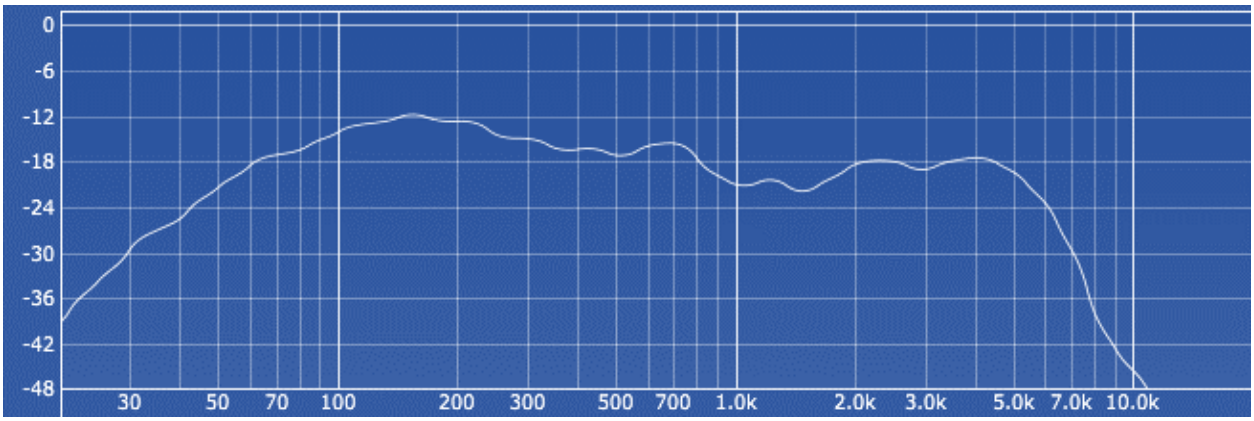
The spike around 650 Hz brings guitars more "forward." It's complemented by a peak between 2 and 4 kHz and a full-sounding low end. The midrange and upper midrange peaks emphasize lead-guitar-friendly frequencies.

4x12 Uber T75



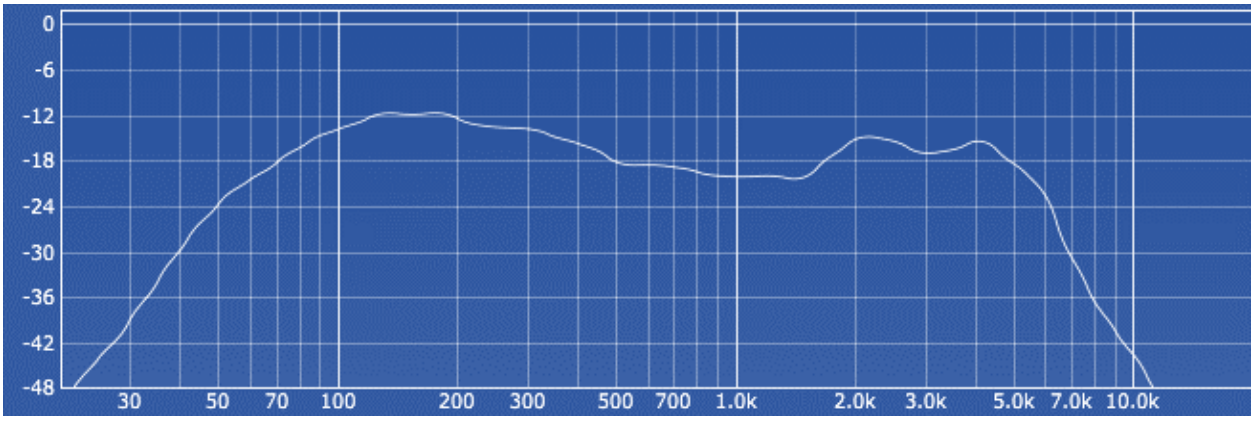
This 4x12's curve is fairly uniform, with a little more midrange scoop than the Brit and Cali V30 cabs. There's also a slight high-frequency resonant peak above 5 kHz, before the high frequencies fall off rapidly.

4x12 Uber V30



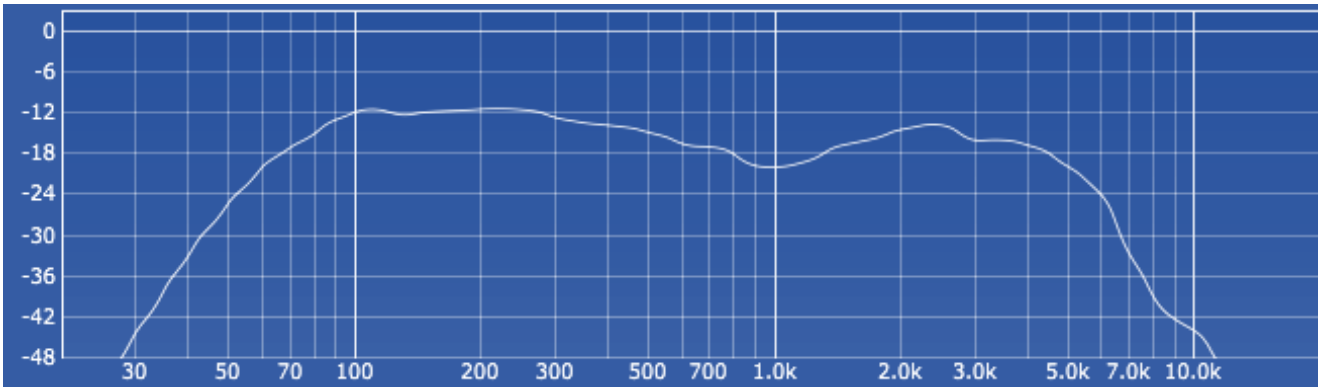
For a 4x12 cab, the response is relatively even. However, like the Moo)))n T75 and Cali V30 cabs, there's a peak around 650 Hz. This adds a resonant quality. The upper midrange from 2 to 4 kHz is not as prominent as some other cabs.

4x12 XXL V30



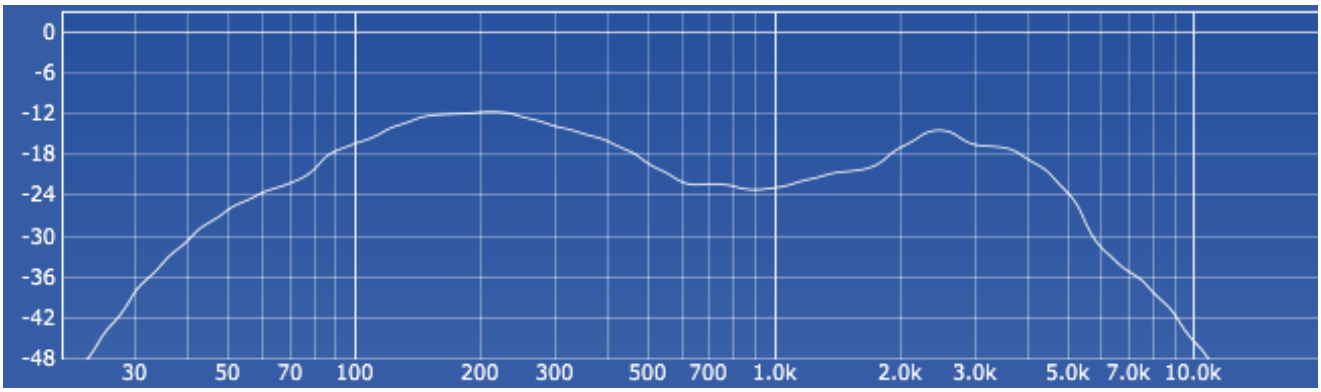
The scoop is milder, and nominally more even, than most 4x12 cabs. Although a lot of cabs have high-frequency lifts, the peaks are more widely separated. Also note there's plenty of bass depth. This is a versatile, general-purpose cab.

4x12 Cartog Guv (1.5)



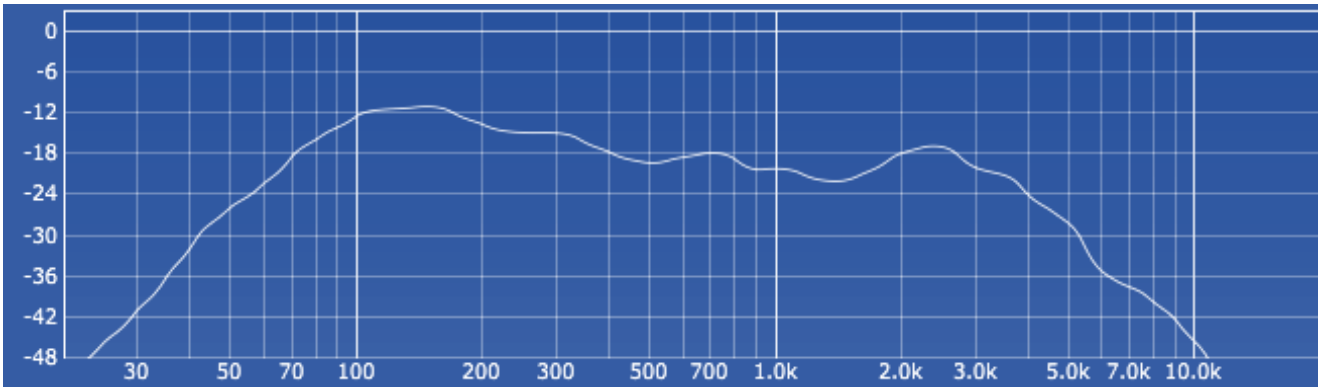
The low-frequency rolloff gives a tight bass. The lower mids are prominent, but slope down gently to a midrange scoop centered around 1.0 kHz. The high-frequency rolloff is typical of 4x12 amps.

4x12 Cartog C90 (1.5)



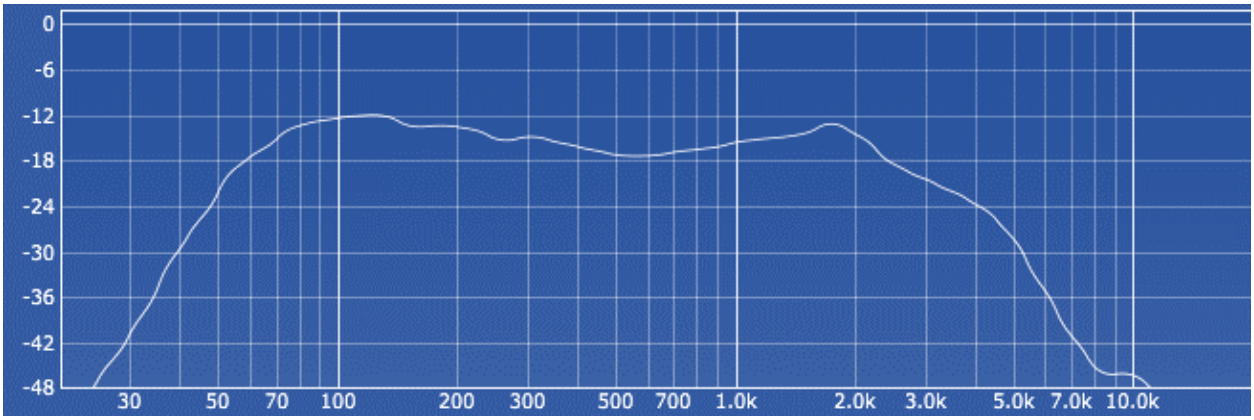
This exaggerates the Cartog Guv's curve. The bass boost is rounder, the midrange scoop is broader and deeper (and slightly lower in frequency), and the peak around 2.5 kHz sounds more prominent than the Cartog Guv because of the scoop and the steeper high-frequency rolloff.

4x12 SoloLead EM (1.5)



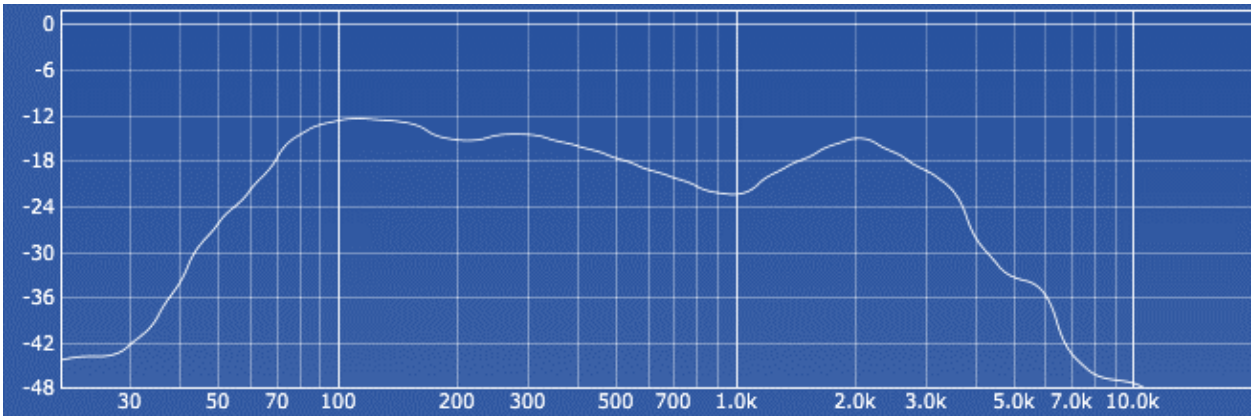
With the prominent low end, extended midrange before hitting a slight scoop, and an articulation peak around 2.5 kHz, this is a good choice for beefy, yet defined, sounds.

1x12 Epicenter



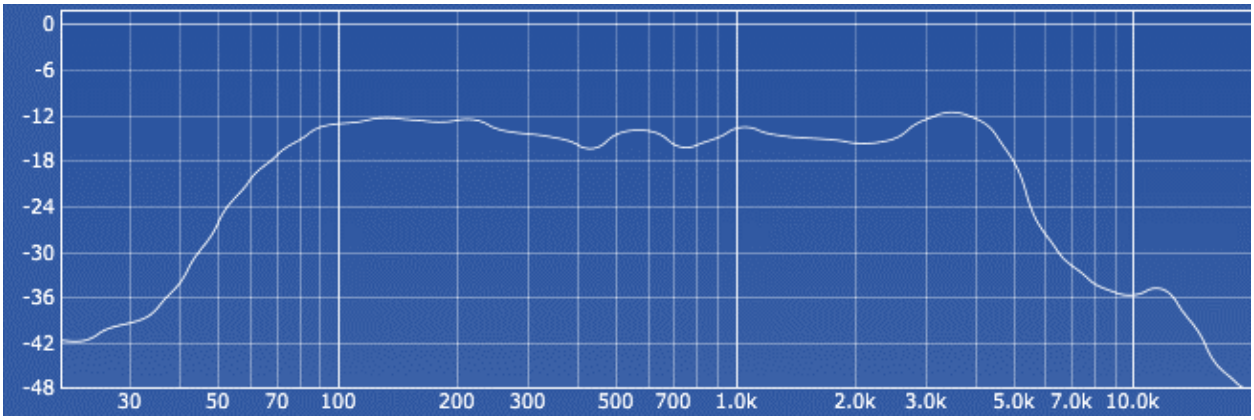
With a mostly flat response, subtle scoop, and slight upper midrange emphasis, this solid, all-around bass cab has a smooth response.

1x15 Ampeg B-15



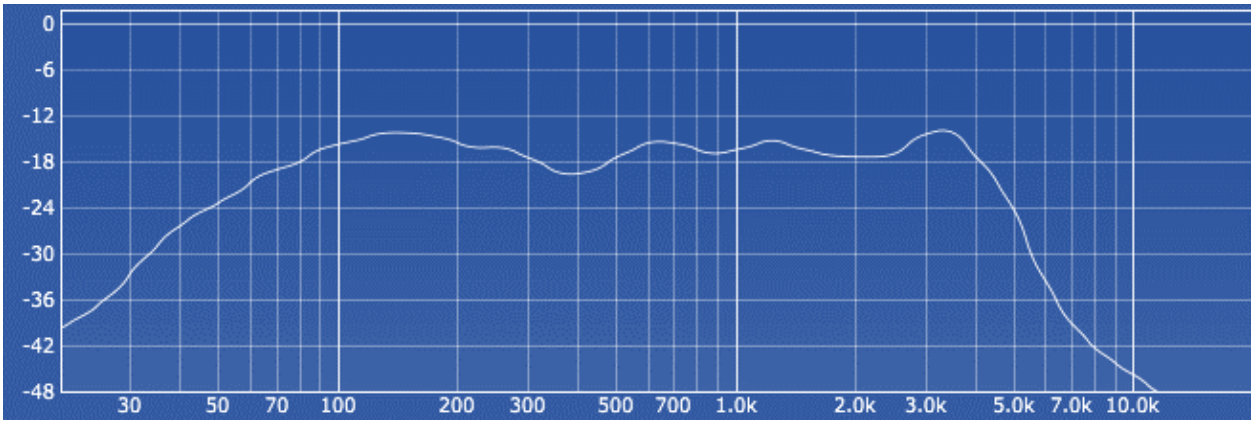
The B-15 has a reputation for a deep, full bass sound with plenty of articulation. This curve shows why: there's a significant bass peak, midrange scoop to make room for other instruments, and a broad peak centered around 3 kHz.

2x15 Brute



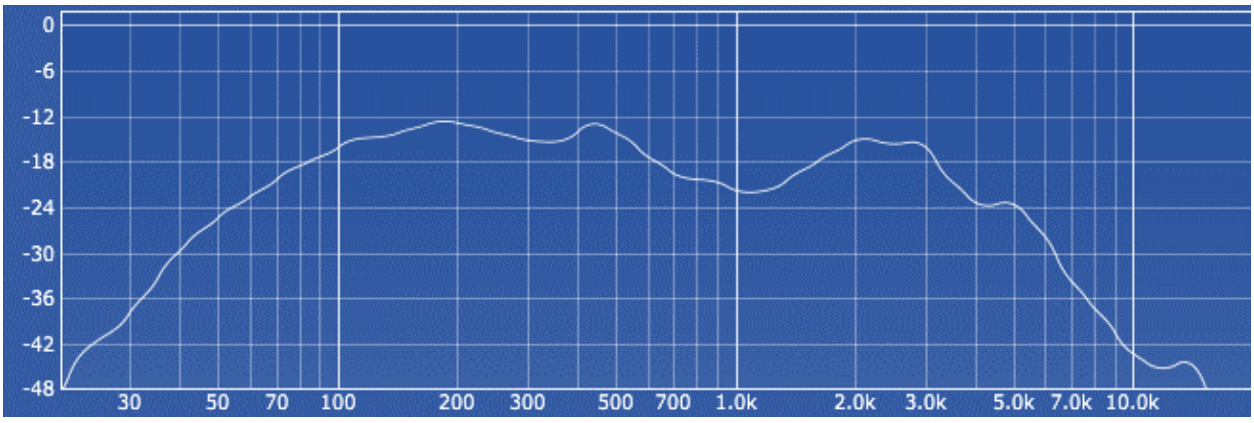
From the midrange on down, the response is relatively flat. This is likely due to the unusual, individually tuned, multi-port system. The upper-midrange peak is from the horn, which contributes the high frequencies needed to emphasize "string zing" and articulation.

2x15 US Dripman



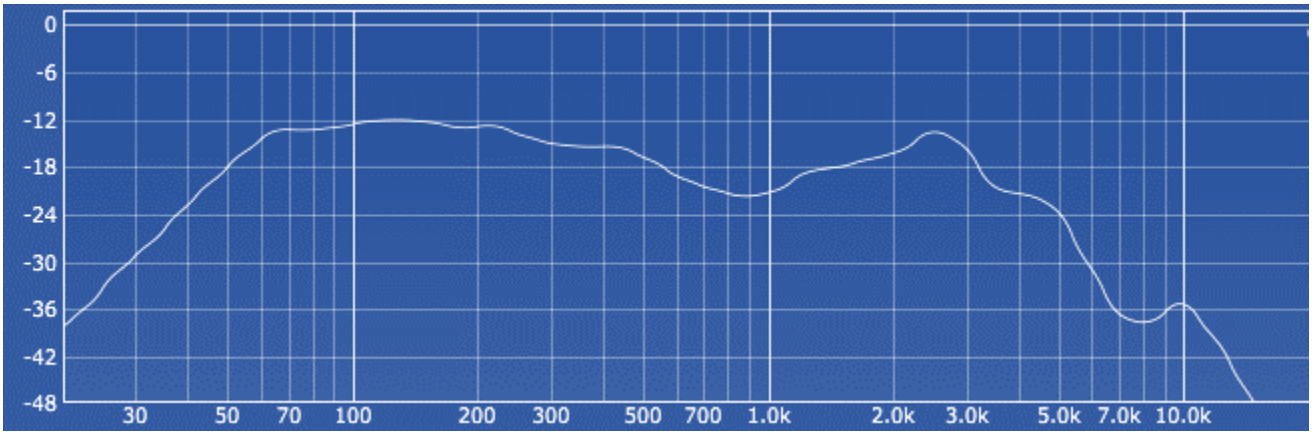
Although nominally a bass cab, many guitar players appreciated the flat response over the guitar's range, with an upper midrange bump for definition. The JBL D130 speakers had decent highs, so there was no horn.

4x10 Garden



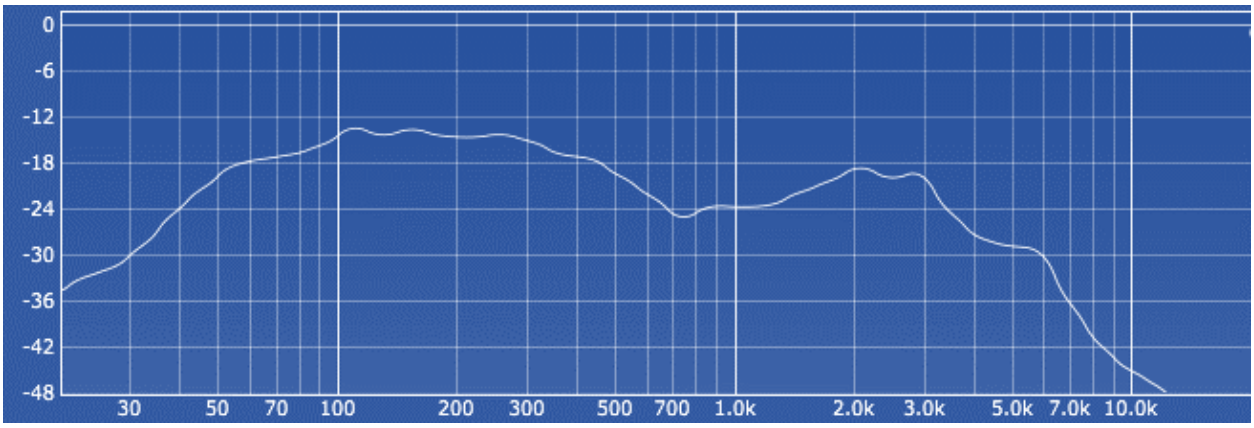
This splits the sonic difference somewhat between the previous two bass cabs. The Garden has more of a scoop than the Brute, and a slightly reduced low end compared to the B-15.

4x10 Ampeg Pro



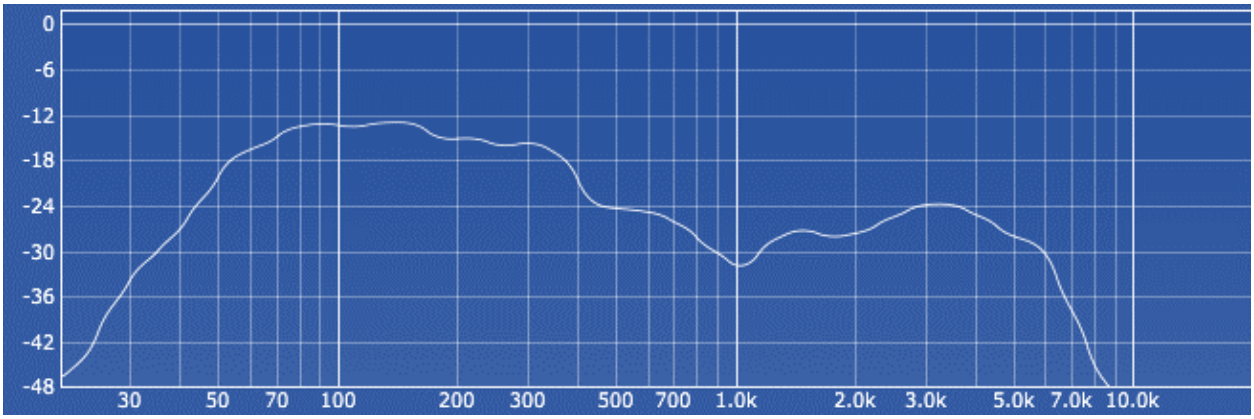
4x10 cabs tend to have more similarities than differences, and 4x10 Ampeg Pro and 4x10 Garden are no exception. However, the Ampeg Pro has a narrower upper-midrange peak, and a peak at 10 kHz for more "air."

6x10 Cali Power



The remaining bass cabs all have horns, which explains the upper midrange emphasis. Without this, the bass would have a hard time sounding defined in a mix. The pronounced scoop makes the highs and lows more apparent.

8x10 SVT AV



This is the other Helix bass cab whose physical counterpart doesn't include a horn. As a result, this one's all about the bass, despite the midrange scoop and a peak around 4 kHz.

Do the Response Curves Actually Matter?

Even the slightest changes to a cab's parameter can change the response. Mic distance has a major effect on the sound, especially with a Dual Cab setup. These parameters also interact. Increasing distance can make the sound brighter. But move the mic position more to the edge, and the sound dials back the brightness.

If you create your own presets, analyzing the curves is useful. For example, suppose you've created a preset where the most important elements are the distortion, amp, and cab. If the sound is harsher than desired, you might consider EQ to reduce the highs. But this new variable will interact with both the distortion and the amp.

By looking at cab curves, you may see a cab with a significant upper-midrange peak—not the distortion—causes the harshness. Then, look at other cab options. Choose one that comes close, but has a more subtle peak.

After a while, you'll be able to anticipate which cab will likely produce the sonic character you want. There are certain cabs that I use a lot, but before I analyzed their curves, I didn't know why. After the analysis, I could see they had certain characteristics in common. To add more variety to my presets, I looked for cabs that were like my favorites, but perhaps had some frequency response variations that added character. These could be useful for other presets that use different effects.

The Modeled Mics: General Information and Parameters

Knowing how to use Helix's mics streamlines the process of optimizing the cab sound. I haven't felt much need for IRs since "cracking the code" for the mics. The new cabs include parameters the Legacy cabs don't have. In addition to new cabs, version 3.50 added 12 new mics for the cabs.

Dynamic Mics	Based on
57 Dynamic	Shure SM5
421 Dynamic	Sennheiser MD 421-U
7 Dynamic	Shure SM7
906 Dynamic	Sennheiser e906
30 Dynamic	Heil Sound PR 30
52 Dynamic*	Shure Beta 52A
88 Dynamic*	Beyerdynamic M88TG
D6 Dynamic*	Audix D6
40 Dynamic*	Heil Sound PR 40
* Bass Cabs only	
Ribbon Mics	**Based on**
121 Ribbon	Royer R-121
160 Ribbon	Beyerdynamic M 160
4038 Ribbon	Coles 4038
84 Ribbon	AEA R84
Condenser Mics	**Based on**
414 Cond	AKG C414 XLS
47 Cond FET	Neumann U47 FET
67 Cond	Neumann U67

The Legacy and Version 3.50 mics have the following parameters in common.

Mic

This is the most important cab parameter, because of the frequency response differences among various mics. Companies "tune" physical mics for specific frequency responses. This can be for subjective reasons (e.g., boost presence for a vocal mic), or to flatten the response as much as possible. Rather than alter a mic's frequency response solely through EQ, you can choose a mic that already incorporates the desired characteristics. Then, you need EQ only to fine-tune the response.

The three most popular mic technologies are *dynamic*, *ribbon*, and *condenser*. Each has its own characteristic sound and frequency-response characteristics. This chapter includes sections on each mic type.

Miking physical guitar amps is an art. Some engineers use one mic on an amp, some two or more, and sometimes, they add room mics as well. With Helix, each cab can have only one mic. However, by using Dual Cabs set to the same cab, you can use two different mics and mic positions, then mix them together. The result is the same as using two mics on one cab.

Room mics capture room reflections. The Legacy cabs have an Early Reflections parameter that emulates these reflections. The version 3.50 cabs do not include this Early Reflections parameter. Fortunately, the Dynamic Ambience reverb processor added to version 3.50 provides a more advanced and flexible implementation for creating early reflections.

Distance

Distance has two major effects on your sound (fig. 6.4).

- Mic distance affects the frequency response of individual cabs. Placing directional mics (e.g., mics with cardioid pickup patterns) closer to a speaker creates a *proximity effect* that boosts the bass. Generally, ribbon mics have the most prominent proximity effect. Omnidirectional mics, whose pickup pattern picks up sounds equally regardless of the angle from which they come, have no proximity effect.
- With parallel cabs, placing two mics at different distances from the speaker cone introduces phase shifts. When you mix the two cabs (especially in mono), these phase shifts create subtle variations in the overall sound.

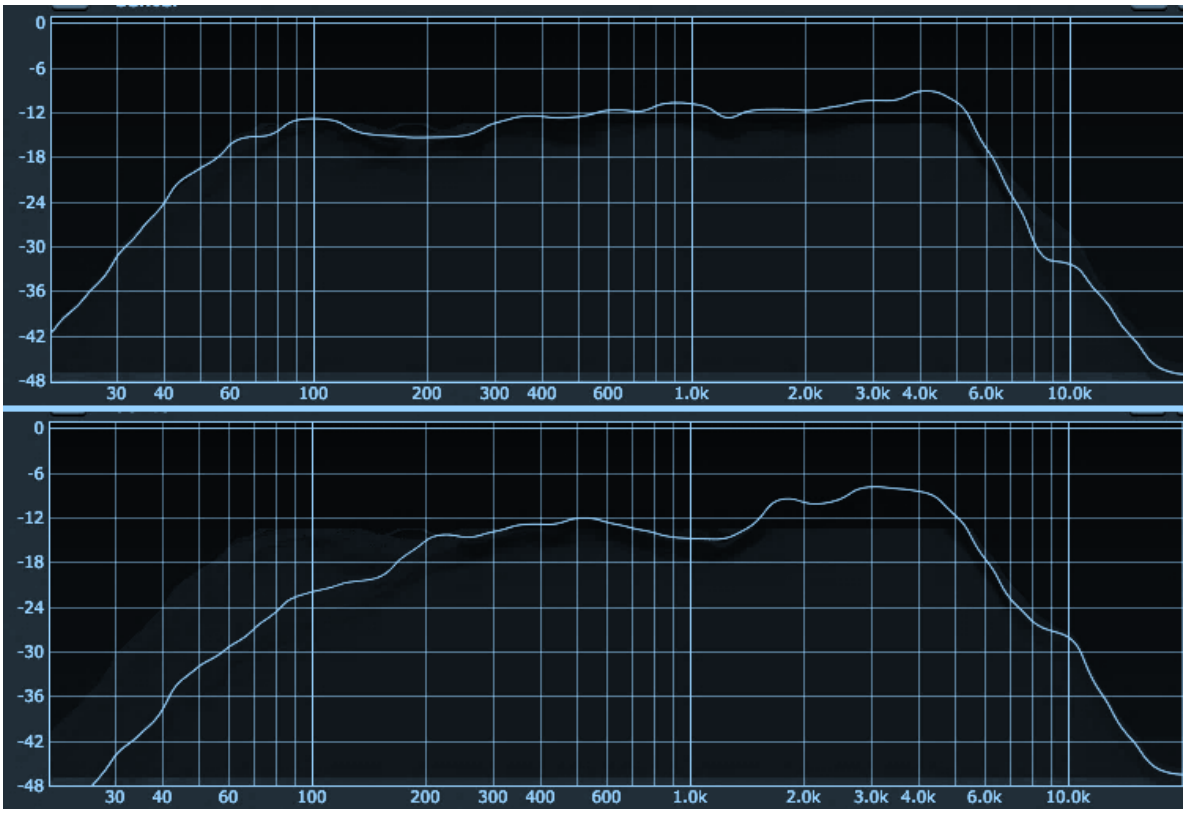


Figure 6.4 (Top) The 30 mic is 1" away from the speaker cone. (Bottom) The 30 mic is 12” away. Note the increased bass due to the proximity effect when the mic is closer to the speaker cone.

Low Cut

This trims the mic and cab’s low-frequency response. Because moving the mic closer to the speaker increases the bass response due to the proximity effect, teaming the Distance parameter with the Low Cut control provides creative options for tailoring the low-frequency response.

Some physical mics include one or more fixed, low-frequency rolloffs to modify the low-end response. The intention is to minimize issues like p-pops and the proximity effect. Rather than model highpass filters on the mics that include them, the variable Low Cut parameter is a more flexible alternative.

High Cut

This trims the mic and cab's high-frequency response. Rolling off highs can reduce harshness and emphasize the mic's low-frequency pickup. Note that all cabs have some high-frequency rolloff, regardless of the mic's High Cut parameters. This is because they model physical cabs, which have a limited high-frequency response.

Mic and Miking Parameters

Compared to the mics prior to version 3.50, cabs in version 3.50 and above include mic placement parameters (fig. 6.5). The new parameters offer more placement flexibility.

Figure 6.5 Version 3.50 Dual Cab parameters. Each speaker has the same set of parameters. As with pre-version 3.50 Dual Cab blocks, click on either upper tab to change the cab type.

Pan

This parameter is available only with Dual Cabs. Each cab can pan anywhere in the stereo field. This is unlike pre-version 3.50 Dual Cabs, which panned the two cabs hard left and right. It's no longer necessary to add a post-cab block to pan cabs more flexibly.

Delay

Like Pan, this parameter is available only with Dual Cabs. The resolution changes for different delay times:

- **0 to 0.5 ms:** Delay time is variable in 0.02 ms increments. This fine-tunes the phase adjustment between two cabs.
- **0.5 to 5 ms:** Delay time is variable in 0.1 ms increments, for coarser tuning when compensating for phase issues.
- **5 to 50 ms:** Delay time is variable in 1 millisecond increments. This allows for creative effects, like placing one speaker further from the other in the Helix's "virtual room."

Position

This parameter moves the virtual mic horizontally across the speaker's front. Placing the mic closer to the edge reduces the high frequencies (fig. 6.6). This takes some "edge" off brighter amps and cabinets. The result is a warmer sound.

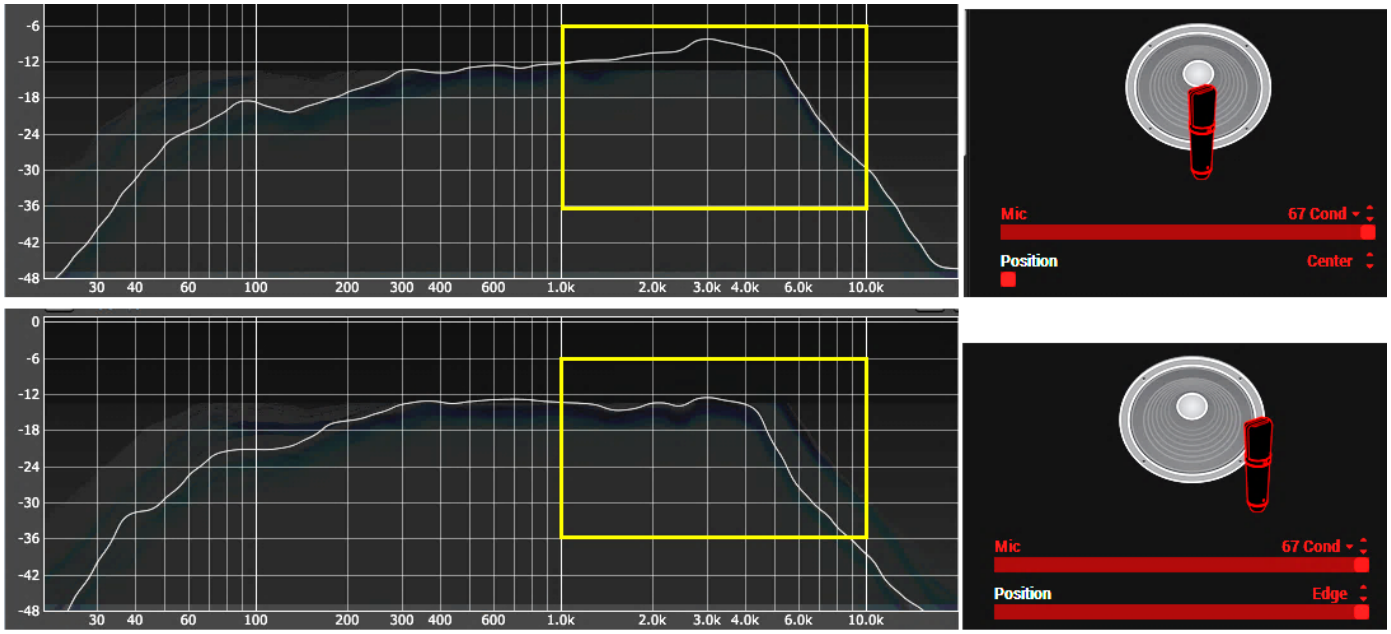


Figure 6.6 The area outlined in yellow shows the high-frequency response with the mic in the center position (top) and edge position (bottom). Both mics are at a 45 degree angle.

Angle

Another version 3.50 option changes the mic's angle in relation to the cone (fig. 6.7). This has a subtle effect with a single cab. The effect is most apparent when the mic is at the edge and positioned close to the speaker, or in a Dual Cab setup. The effect is less apparent as the mic moves further away from the speaker.

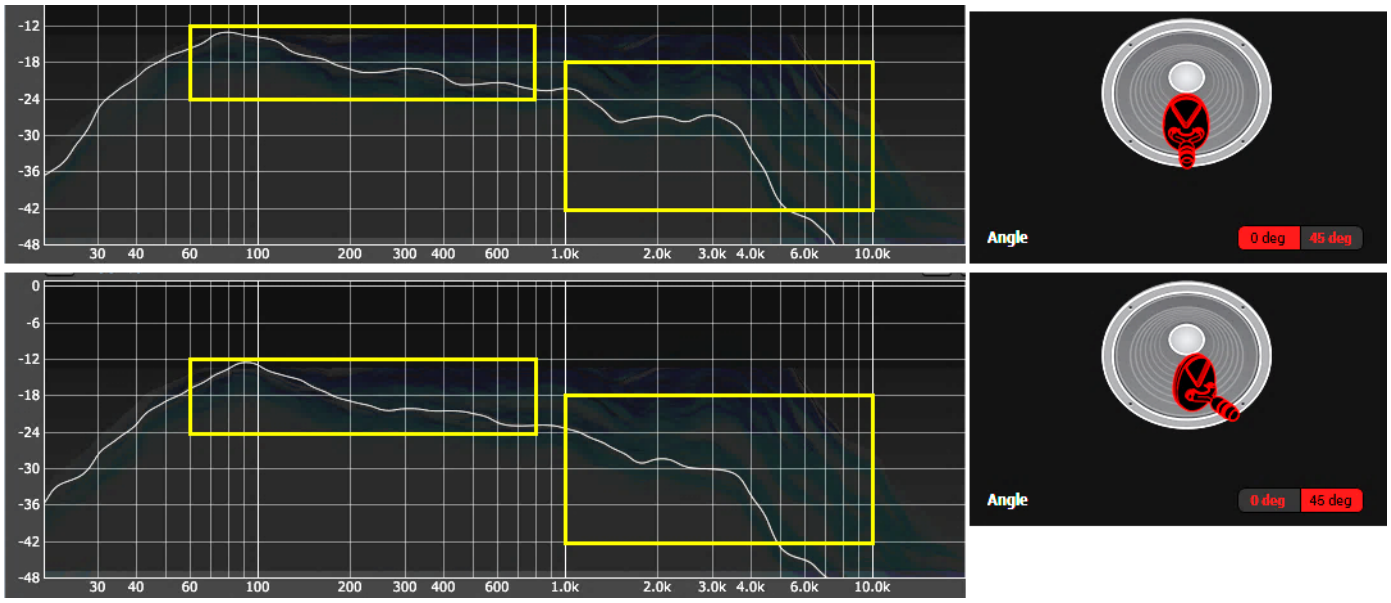


Figure 6.7 Changing the 4038 mic's angle from 0 to 45 degrees creates various response differences.

Legacy Cab Mic Parameters

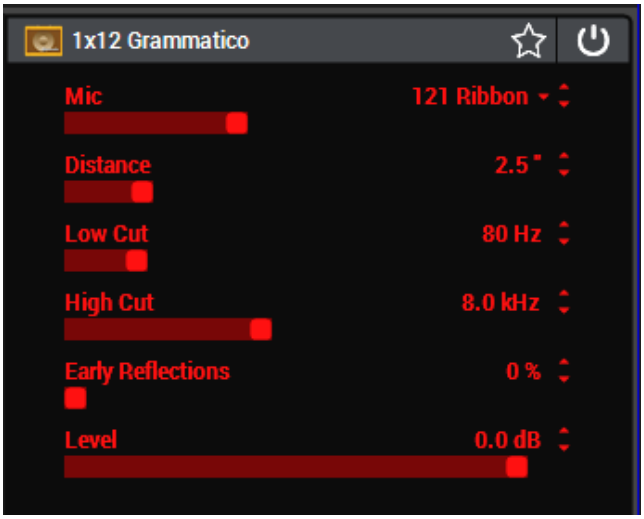


Figure 6.8 Except for Early Reflections, the Legacy cabs include parameters common to the version 3.50 cabs.

Early Reflections

Early reflections (fig. 6.8) add a sense of space, by bunching together tight echoes that are characteristic of sound bouncing around in a room. These persist to a maximum of about 0.3 seconds, but such a short delay doesn't sound like what we usually associate with an "echo effect" (fig. 6.9). You probably won't notice the reflections much past 200 ms.

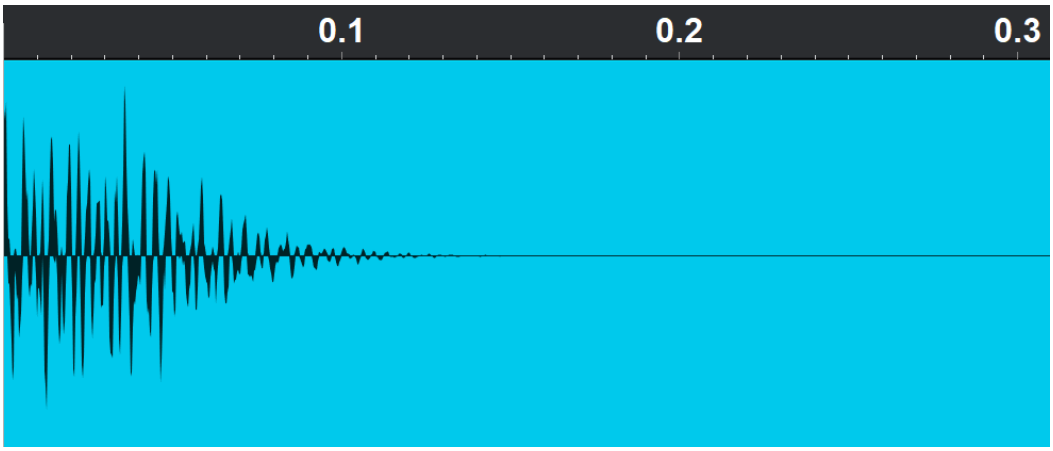


Figure 6.9 This early reflections waveform persists after a pink noise input drops abruptly from maximum level to 0. The upper calibration is in seconds.

The tone of the reflections varies with the cab, because the cab's sound is being reflected. Different mic distances create slight tonal differences, but these probably won't be noticed in the context of a mix.

Emulating Mic Edge Placement with Legacy Cabs

Legacy cabs can't move the mic toward the speaker's edge. However, the Tilt EQ can create an effect like moving the mic horizontally across the speaker. Moving the Tilt parameter toward the left (fig. 6.10) reduces the high-frequency response, so the sound is more like placing the mic at the speaker's edge.

Figure 6.10 The Tilt EQ can approximate some mic placement changes with Legacy cabs.

Panning with Legacy Dual Cabs

With Legacy Dual cabs, one cab is panned hard left, and the other, hard right. However, you can modify panning in the post-cab signal flow.

To alter the hard right/left stereo imaging, insert a Stereo Volume/Pan block after the Dual Cab block. Choose the Stereo Width parameter (fig. 6.11).

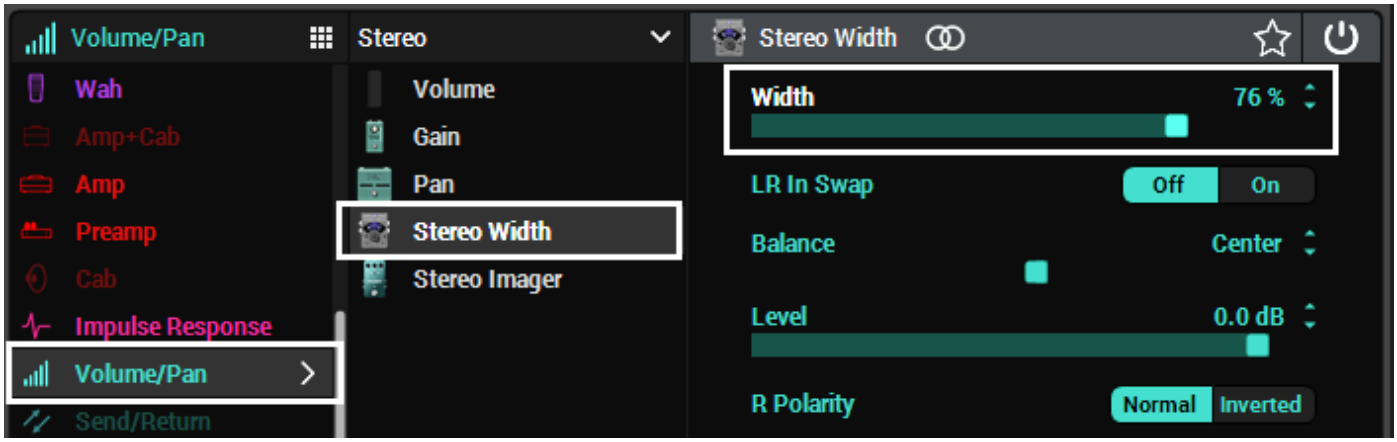


Figure 6.11 With a Legacy Dual Cab block, a stereo Volume/Pan block's Stereo Width parameter can edit the panning width. For example, the settings outlined in white bring the two cabs 24% closer to center.

The Width slider goes from centered (slider full left) to hard left/hard right (slider fully right). In between those extremes, the width increases as you move the slider more to the right. To have the sound "lean" more toward one side of the stereo image than the other, edit the Balance parameter.

This technique is useful for presets that serve double duty for stage and studio. In the studio, where a wide stereo image may be desirable, increase the width. For the stage, narrow the image so that it's closer to mono.

Tip: Changing **R Polarity** from Normal to Inverted may sound good in stereo because it widens the image, but if played back in mono, there will be at least some cancellation between the two channels.

Current and Legacy Mic Responses

This section compares version 3.50 and Legacy mics. These tests are *not* super-lab-quality accurate! However, because all the mics were tested under identical conditions, you can make useful comparisons.

Overall, the Line 6 emulations are consistent with published specs for the mics being modeled. To understand how to get the most out of these curves, please read the following.

About the Version 3.50 and Legacy Mic Test Setups

The Helix mics can't be evaluated separately from cabs, so they can't be compared to an absolute standard. This required choosing a standard test setup. Then, the mics could at least be compared to each other.

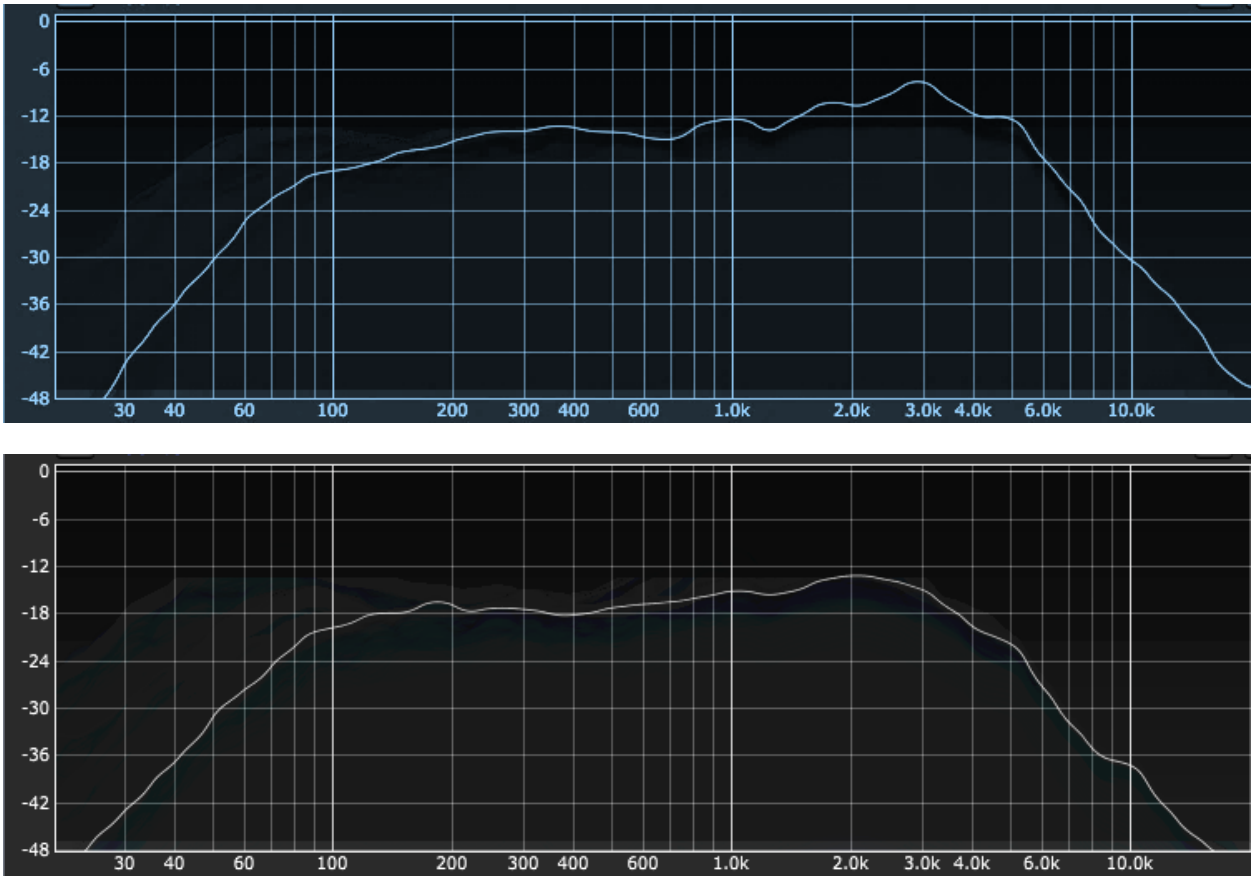
- Some mics from pre-3.50 versions are duplicated in 3.50. **Legacy mic graphs are black and white. 3.50 mic graphs are tinted blue.**
- Version 3.50 mics test parameters were: Version 3.50 Princess 1x10 cab, 6" Distance, Center position, 0 Degree angle, and Low Cut and High Cut filters set to Off.
- Legacy mics test parameters were: Legacy Princess 1x10 cab, 6" distance, 0 Early Reflections, and Low Cut and High Cut filters set to off.
- Some Legacy mic curves in previous eBook versions may differ somewhat compared to the curves in this version. The testing procedure has been improved (there's better resolution on the low frequencies, so the response goes down further), and there are some differences between the old and new Princess cabs. Also, the Legacy mics were tested at 6 inches for consistency with the 3.50 mics, instead of the original 5 inches. However, the basic responses are still quite close.
- Some frequency response variations are caused by the 1x10 Princess Cab, not the mics.
- Please remember, these measurements highlight *relative* differences among the mics, not their *absolute* frequency response. [Spectrum analyzer software credit: SIR Audio Tools. The company offers both free and paid versions, as well as other software programs.]

Tip: If a preset has an unused block, I'll often insert a separate amp and cab instead of an Amp+Cab. The miking options are an important part of the new cabs. Some mic settings seem particularly well-suited to certain cabs, so it makes sense to save those cabs as Favorites. Trying that cab with other amps sometimes gives the same magic results.

Dynamic Mics

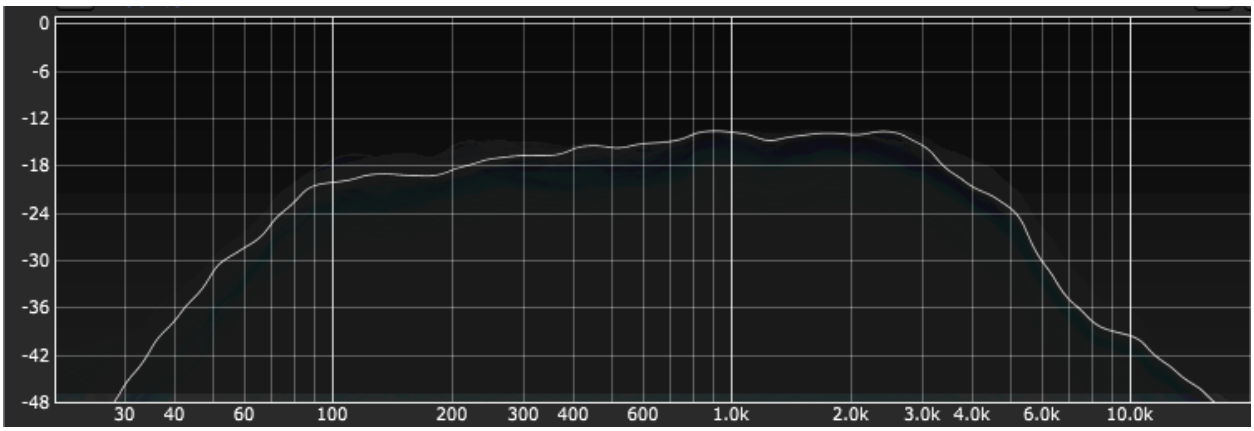
Compared to condenser mics, dynamic mics generally don't respond as quickly to transients. So, some people think they're not as bright, because they lack a bright initial transient. However, dynamic mic manufacturers may tweak the frequency response for a high-end lift. Physical dynamic mics handle high sound-pressure levels and are rugged, so they're popular for live use.

57



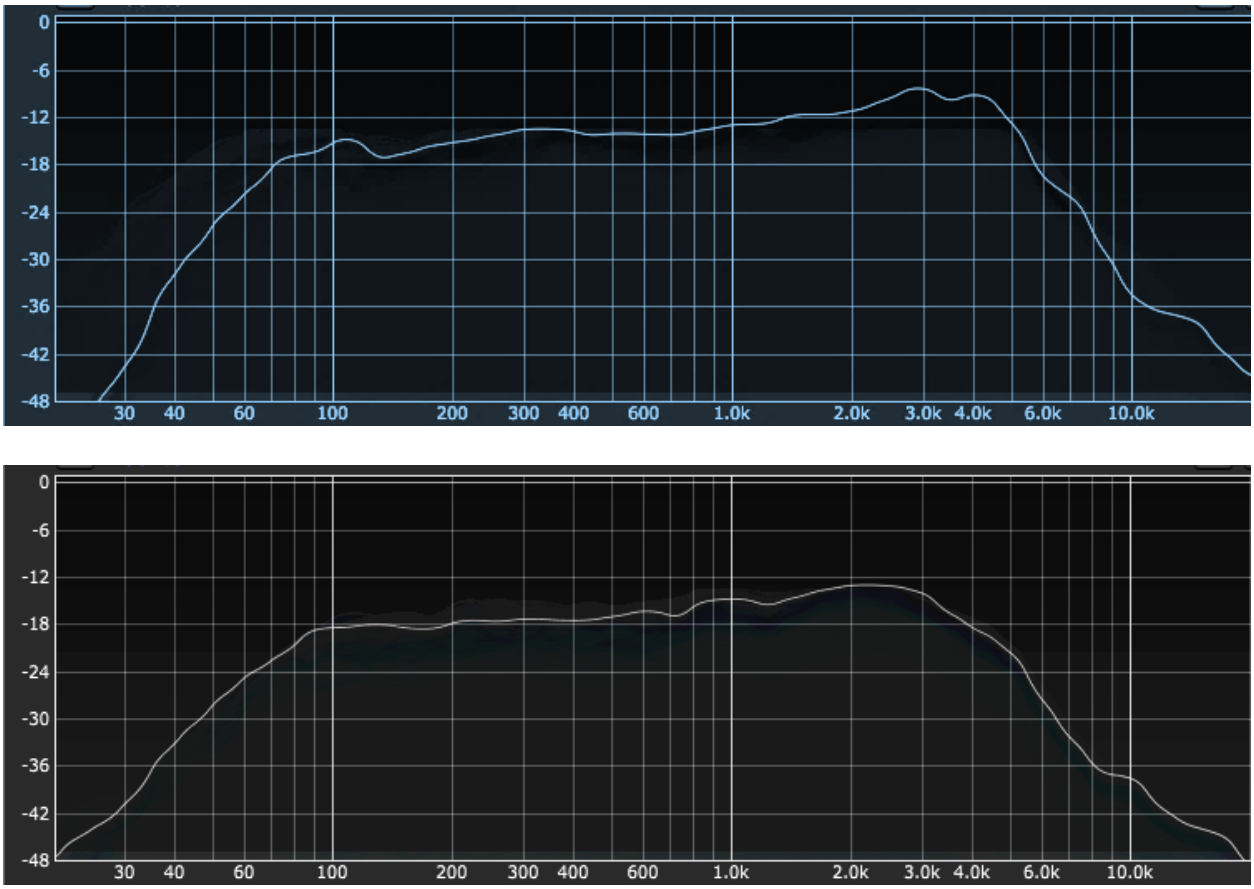
This models one of the most popular mics for guitar amps. It rolls off gently below around 200 Hz and has an upper-midrange emphasis. The frequency response graph correlates closely with the published spec.

409



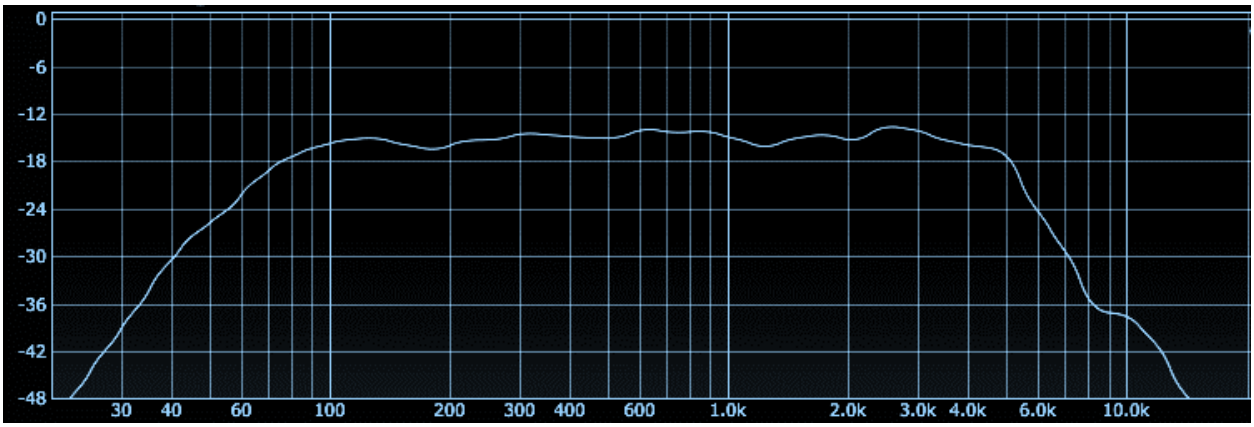
The 409 has a slight high-frequency rise above 200 Hz. As with the 57 and other directional mics, moving the mic closer to the speaker cone increases the bass response, due to the proximity effect. If you don’t need a lot of natural bass, this mic is a good choice for accurate response with a dynamic mic character.

421



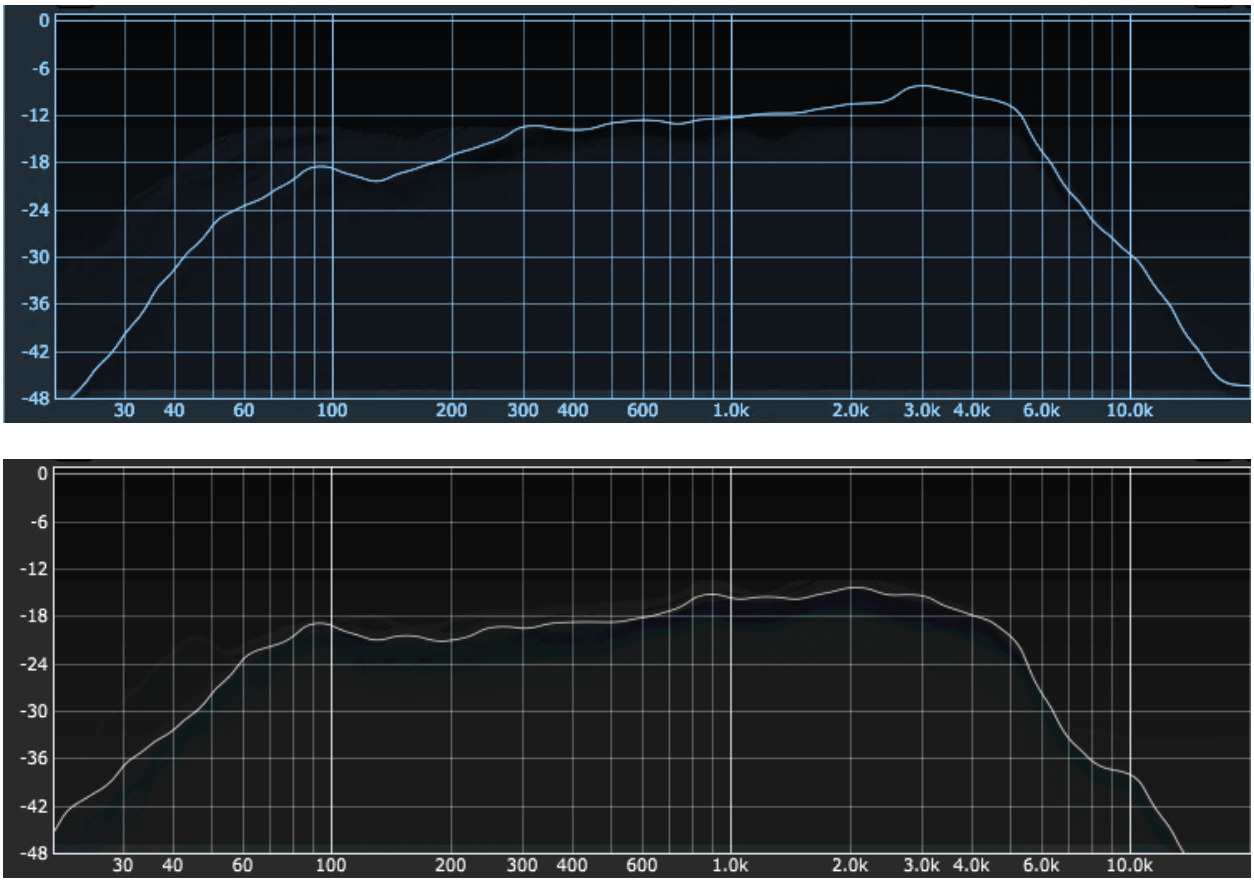
Like the 57, the 421 has a high-frequency lift, but it starts at a slightly lower frequency and covers a wider range. It also goes lower in the bass range, because of its shallower rolloff slope. This dynamic mic's character has good presence, and a bit more low-end response than average.

906



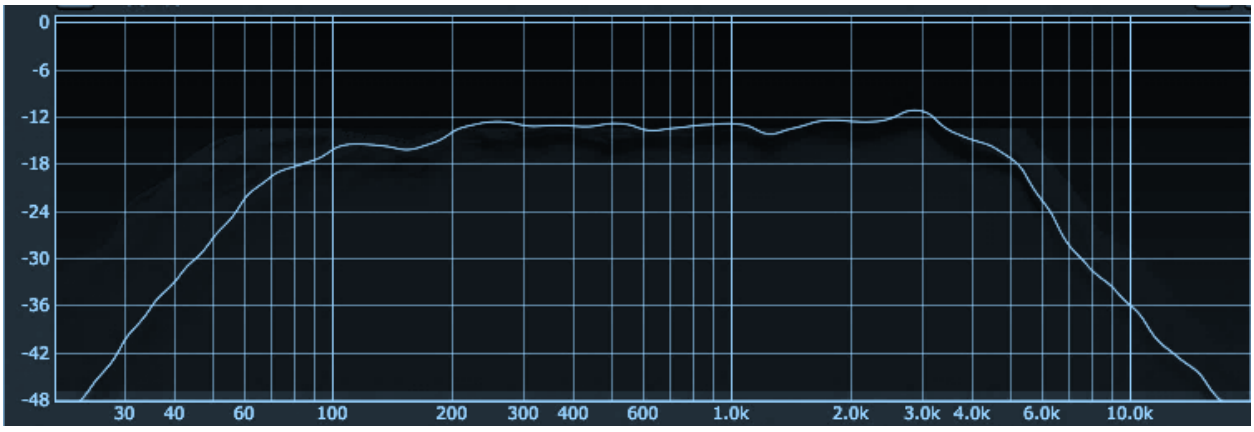
The e906 was designed specifically for miking guitar amps and cabs. The flat front accommodates being placed right up against a speaker. Its supercardioid directional response has made it popular for live use, because it better rejects other sound sources. The Helix model seems to emulate its 3-way presence switch set for the flattest response, which engineers commonly use to reduce harshness with amps.

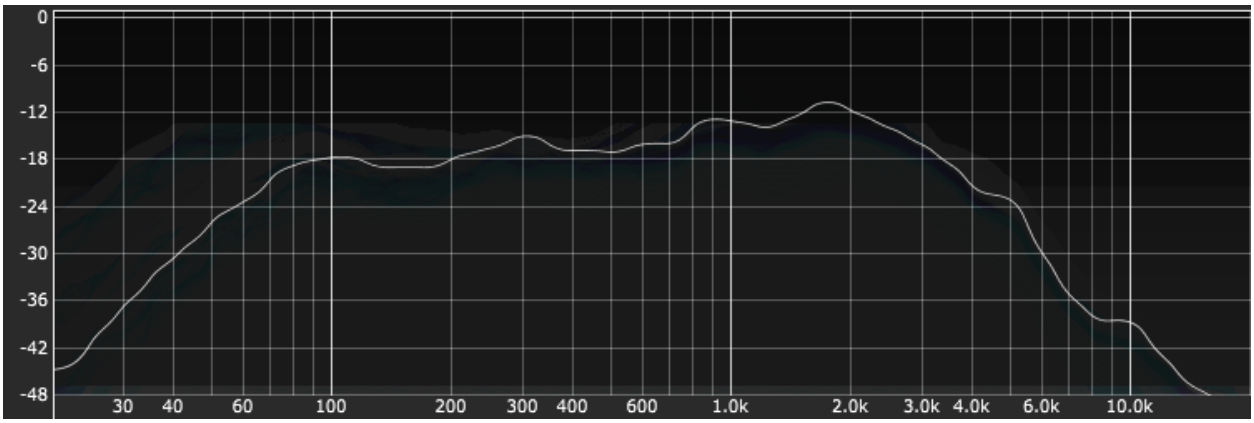
30



The 30 has significant boosts around 80 Hz and 4 kHz. The naturally “scooped” sound has presence where the ear is most sensitive. Use this mic to lift the guitar out of the mix and have an added bass presence, while leaving some space in the midrange for other instruments. The proximity effect is particularly prominent around 80 Hz. It’s more pronounced with the 30 than with the Helix’s other dynamic mics.

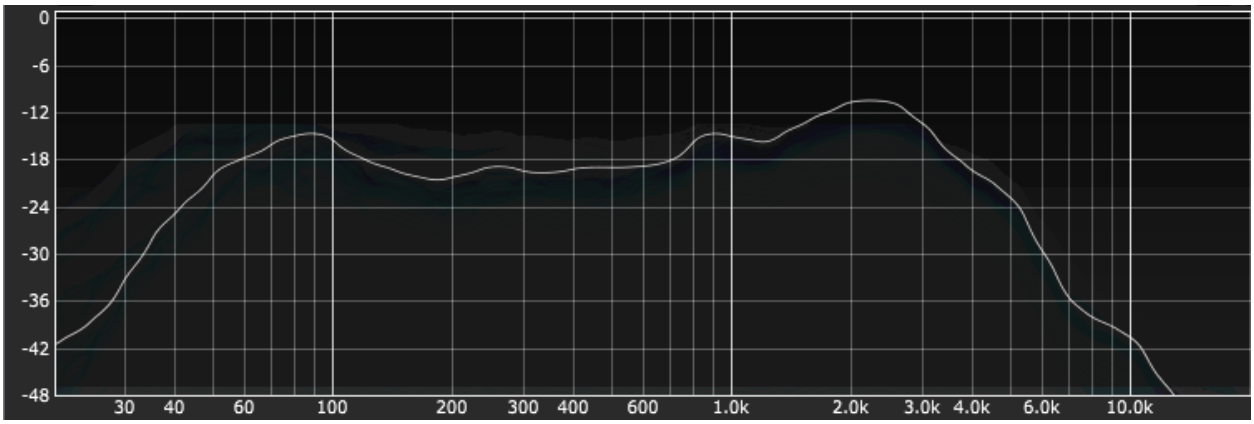
7





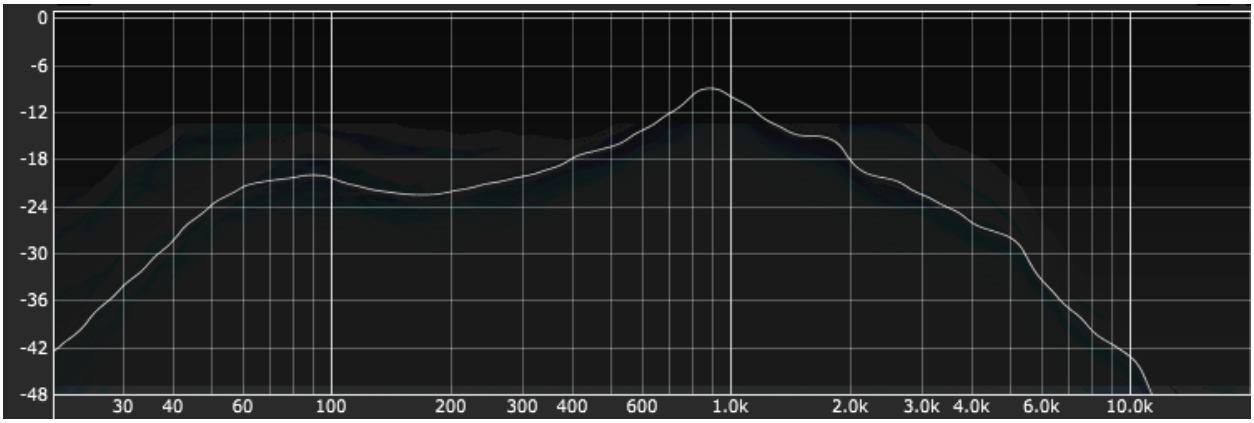
This large-diaphragm mic has been a favorite for broadcasters, podcasters, and streamers. Part of the reason is the consistent response. It also handles the upper midrange well, which improves intelligibility. The physical 7 mic has several peaks and dips in the 4 to 10 kHz range, which you don't see in this averaged response.

112



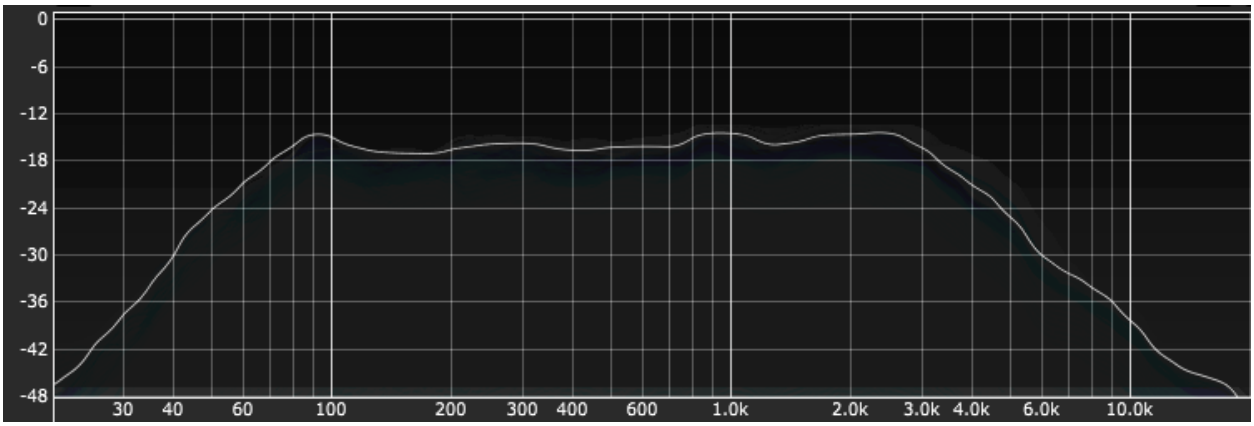
Hey bass players—this dynamic mic's for you! Described as a kick drum mic, its response goes down to 20 Hz. Check out the major low-end boost around 60 Hz (and that's without any proximity effect). The bump in the 2 kHz range is perfect for giving bass extra articulation.

12



Here's the 112's predecessor. The 12 dynamic mic's low-frequency response isn't as prominent as the 112, but the big midrange peak is good for articulation.

20



The 20 is a large-diaphragm dynamic mic, whose transient response is close to that of condenser mics. There's a reasonably flat frequency response, except for a bit of a boost around 100 Hz, 1 kHz, and 2.5 kHz. The bass is more extended than some mics, but the 20 was intentionally designed to give less of a proximity effect. The Helix model reflects this (and I'm impressed that Helix goes to this degree of detail). Traditionally, the 20 has been popular in voiceover and announcer applications, possibly because the reduced proximity effect gives announcers more leeway in how close they are to the mic. This is a good choice for Dual Cab setups where you want to close-mic one cab, without having excessive bass from the proximity effect.

Ribbon Mics

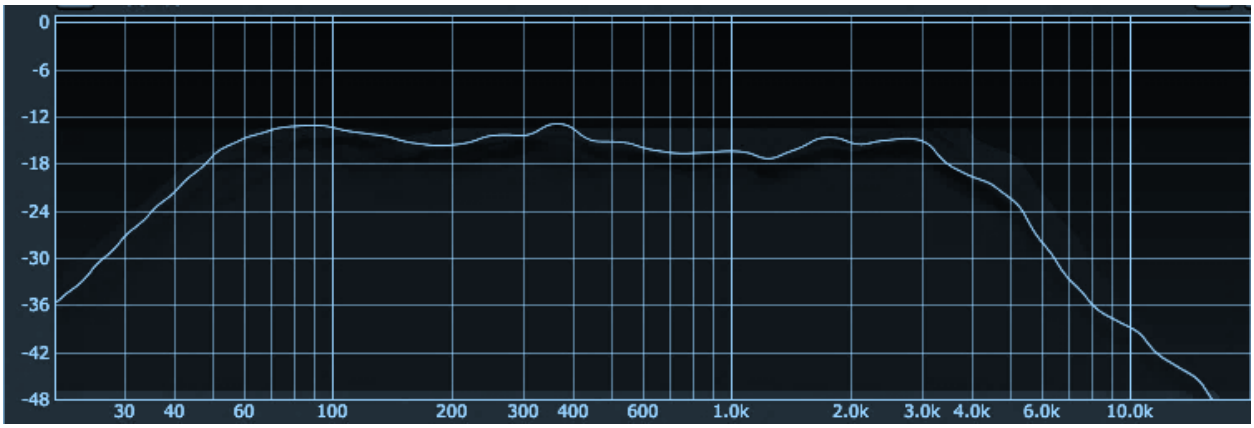
This mic technology offers:

- Excellent transient response
- A warm and distinctive low end
- With some models, reduced high-frequency response in comparison to the bass response
- A smooth-sounding character

Early ribbon mics were fragile and expensive. Newer technologies have led to more rugged mics. Lower production costs have also made ribbon mics more affordable. So, ribbon mics have become increasingly popular for miking amps. You still need to be careful with physical ribbon mics (e.g., don't blow on them!), but that doesn't matter in Helix's virtual world.

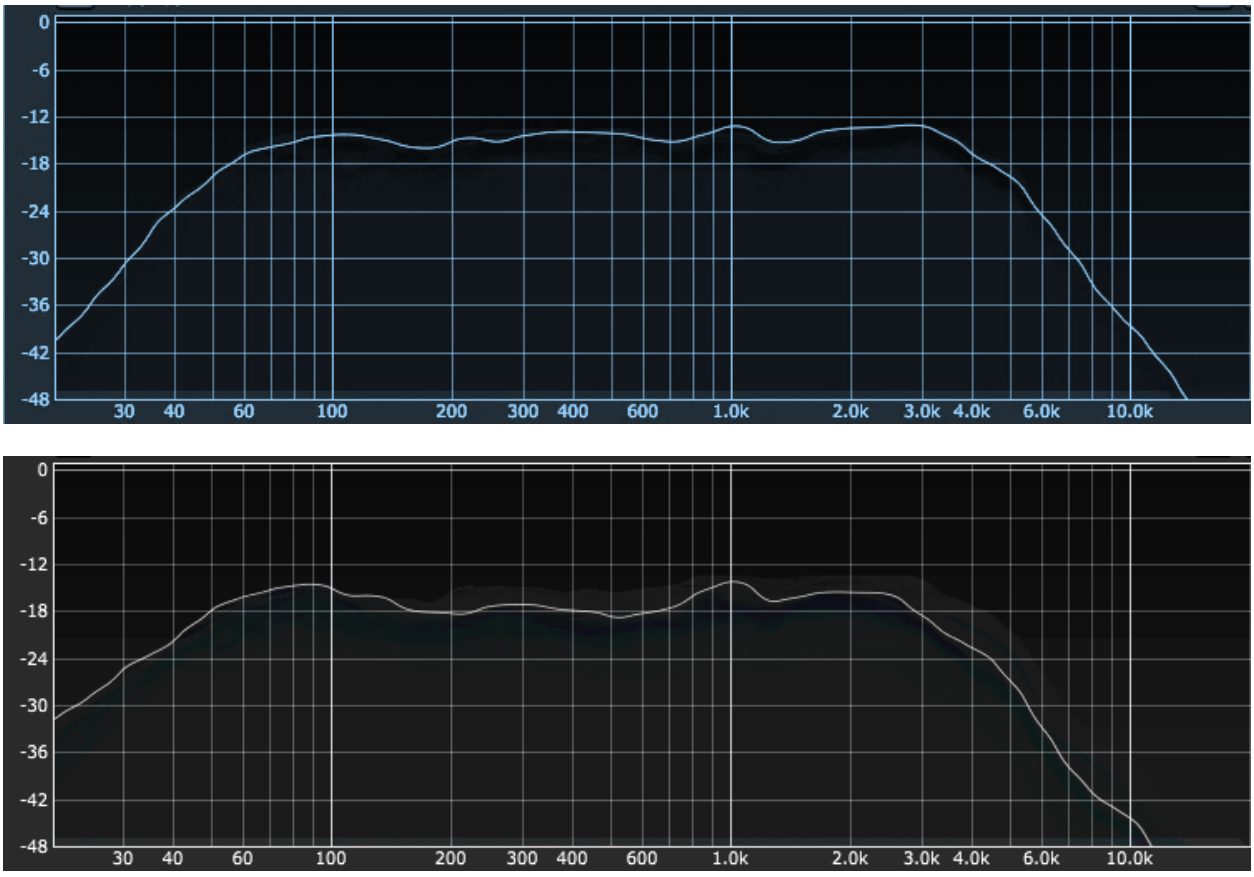
Ribbon mics have a figure-8 pattern, which picks up sounds from the front and back. Rejection of sounds coming from the sides is almost perfect. Because of this, in studios they're commonly used for recording stereo signal sources. Ribbon mics are also excellent for recording duos sitting opposite from each other, because the pickup pattern attenuates what's to the sides.

84



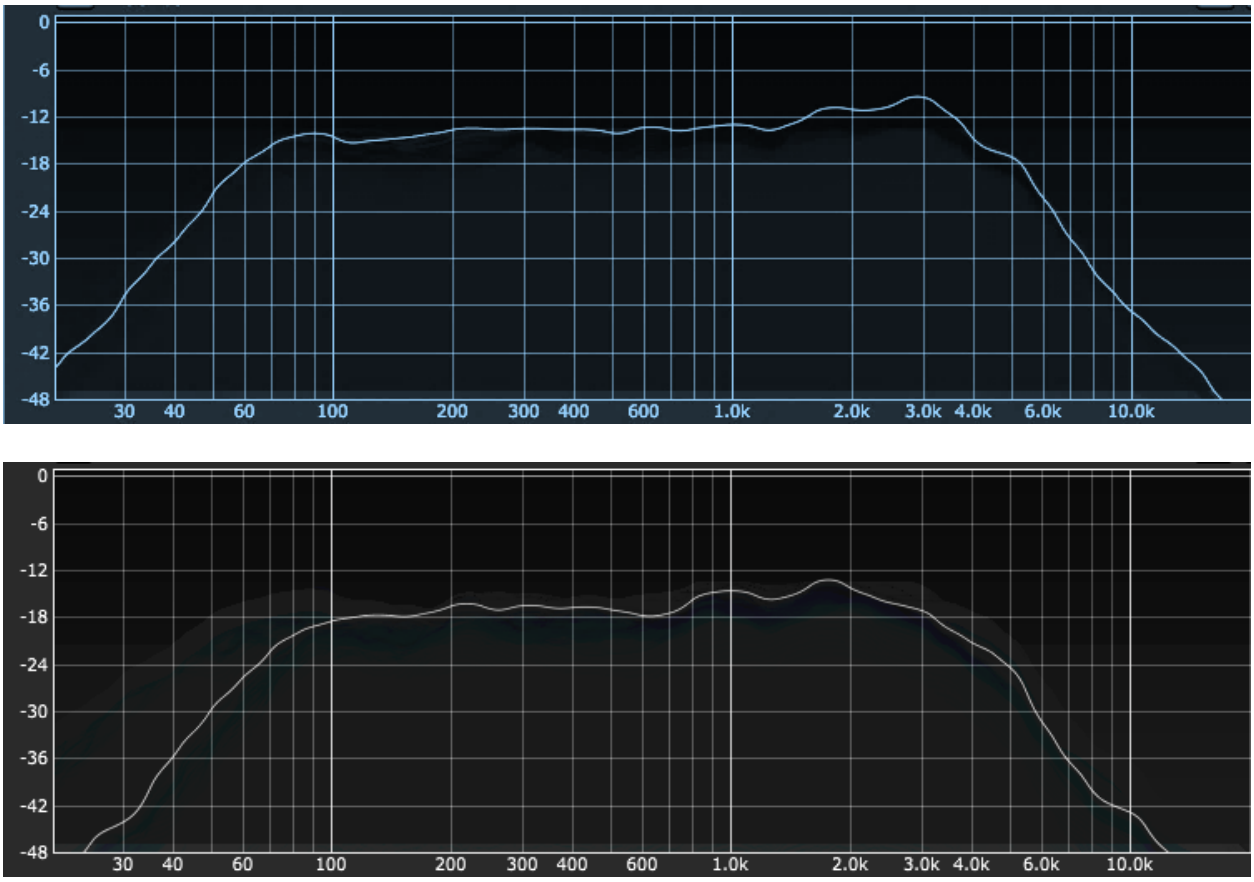
Don’t confuse this with the Legacy KM84 condenser mic model. A descendent of the classic R44 ribbon mic, the 84 model has less proximity effect but retains the characteristic ribbon warmth. It also has a slightly faster high-frequency rolloff compared to the version 3.50 models of the 121 and 160 ribbon mics.

121



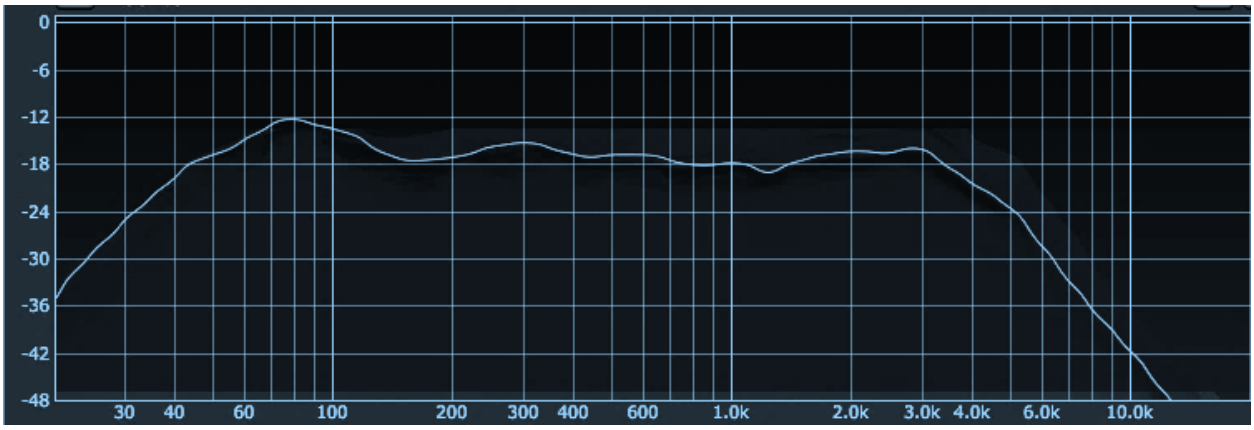
The version 3.50 121 modeling seems closer to the physical mic than the Legacy version, thanks to the extended high end and reduced midrange bulge. The 121 has a crisper response than dynamic mics but not as crisp as condenser mics. It’s a good midpoint between the two. One advantage of a virtual mic is that you can take advantage of its proximity effect, without concern about damaging the mic by being too close to the speaker. Because of the warm sound and solid bass, this mic is a good choice for taking the harshness out of amp sounds.

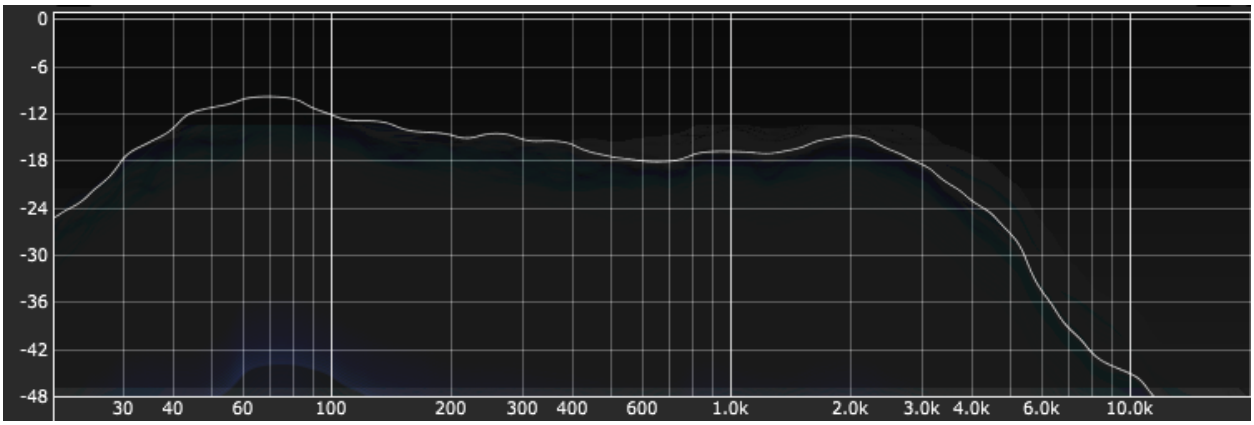
160



This is a consistently popular ribbon mic choice for miking amps. It has a slight upper midrange lift, without the additional low end associated with ribbon mics (although there is a significant proximity effect around 100 Hz). The flatter bass response implies that the mic was modeled around a meter or so from the sound source. Like most ribbon mics, the response falls off at higher frequencies, which can tame amp harshness.

4038



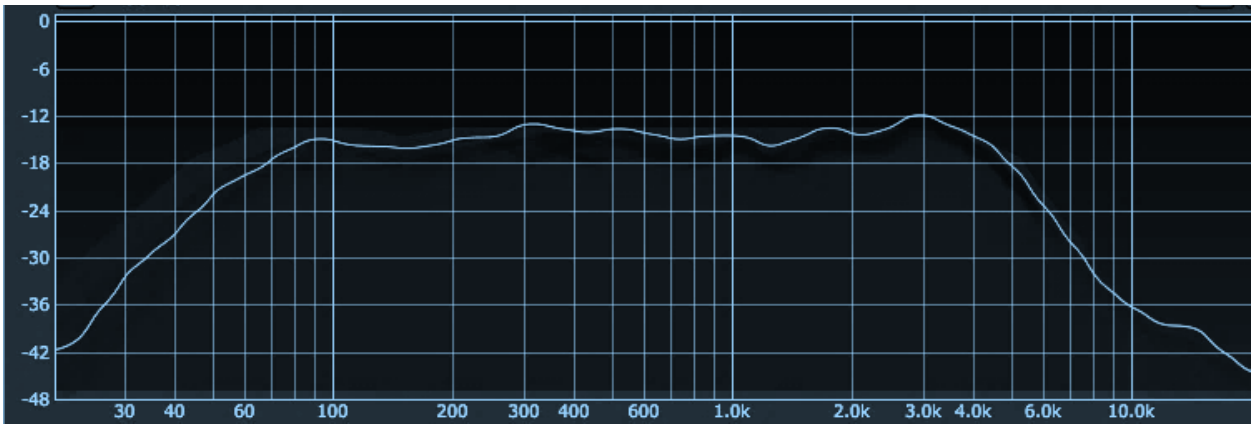


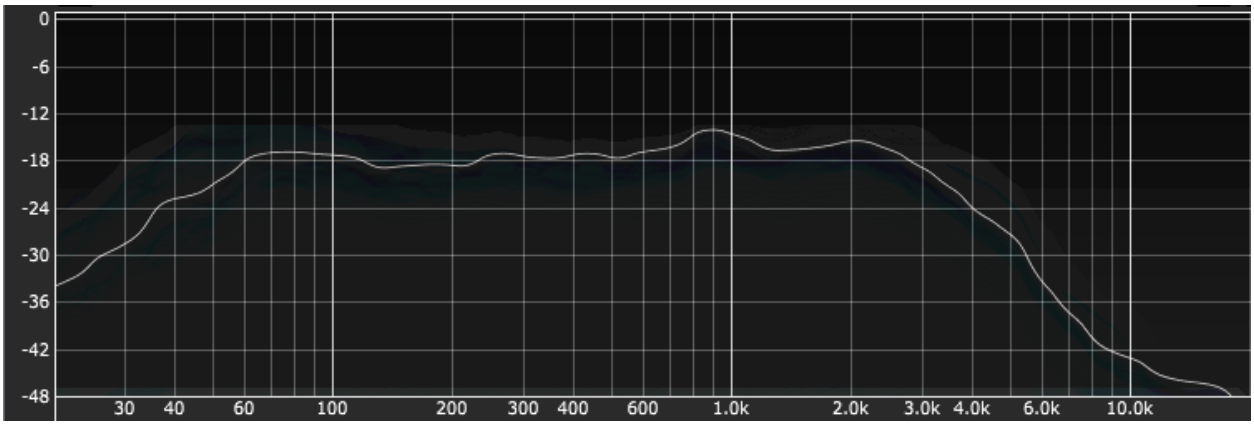
Originally designed and built for the BBC in the 50s, the 4038 has a dominant bass response, the usual high-frequency rolloff associated with ribbon mics, and a slight upper midrange lift. It also has a substantial proximity effect. If you've tried this mic but thought it was too bassy, move it back a bit from the speaker. Moving the mic this way also scoops the midrange more (which happens with many mic models).

Condenser Mics

This technology features fast transient response and good sensitivity. They're not as rugged as dynamic mics and generally require a +48V power source, so they're not as common for live use.

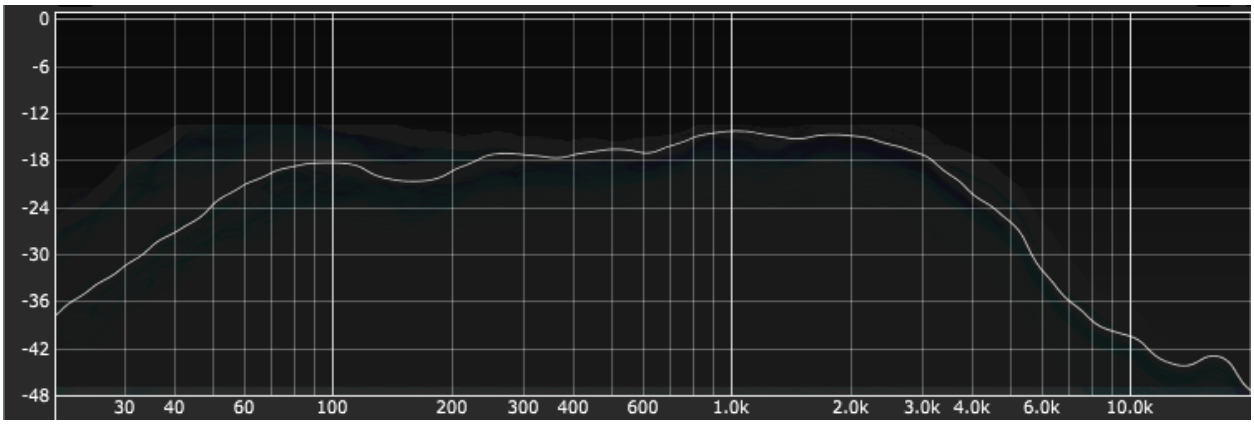
414





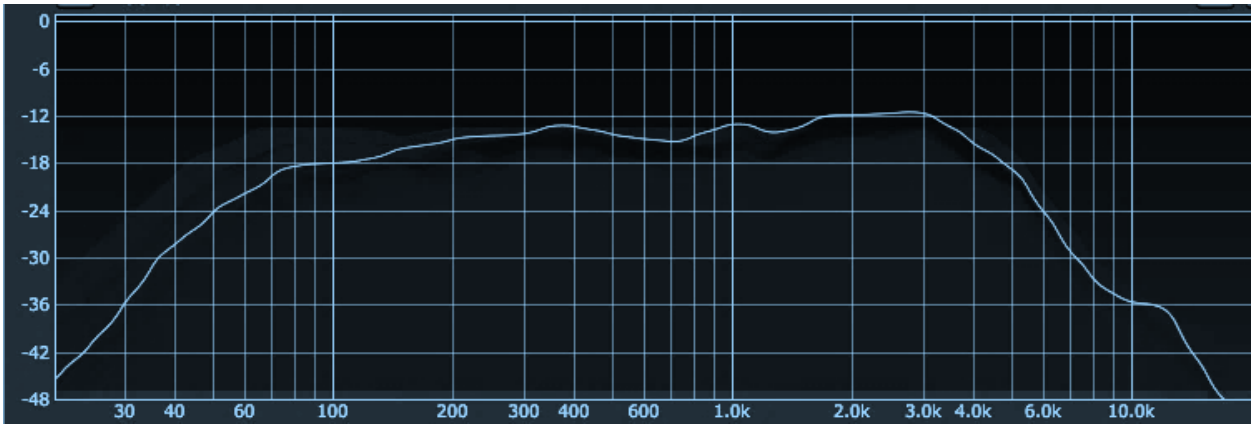
The physical version of this large-diaphragm condenser mic has several polar patterns. The omnidirectional pattern is the flattest, so I suspect that's the mode Helix models. Omnidirectional mics are less prone to the proximity effect. With this mic, it's almost non-existent. The 414 is a fine choice for an accurate, flat response, particularly when close-miking a speaker.

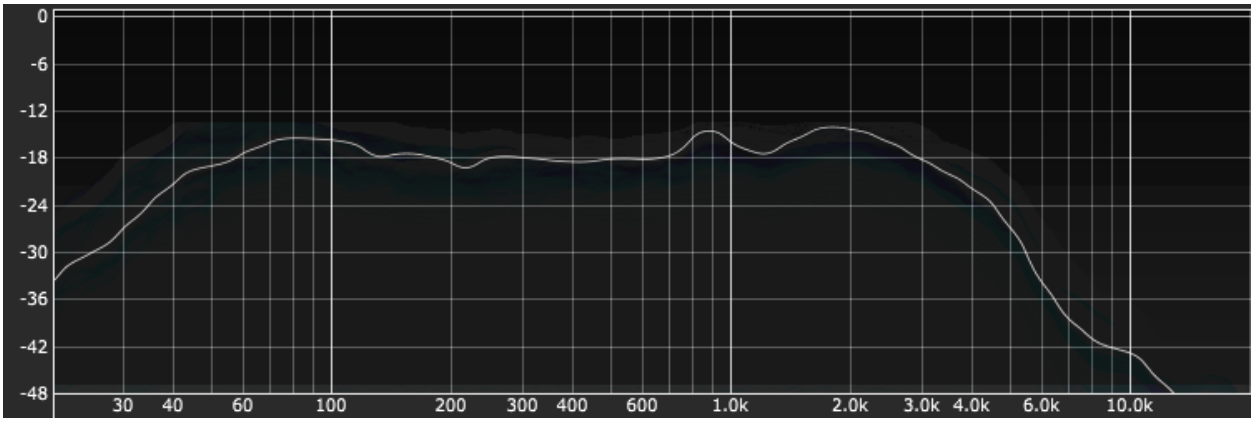
84



A legendary mic with a reputation for a smooth response, this model has a slight midrange lift.

67

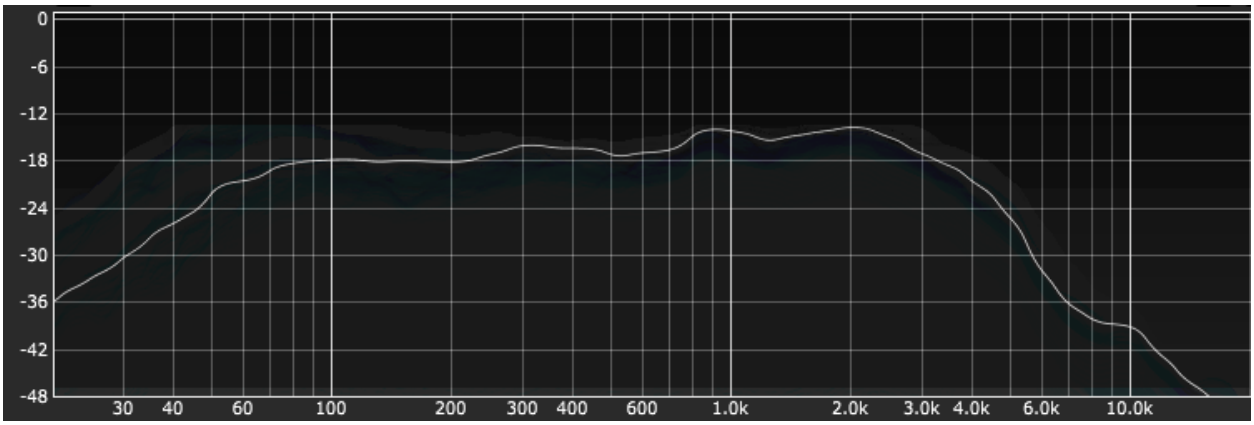




This multi-pattern (omni, cardioid, figure-8), large-diaphragm tube condenser mic includes an audio transformer. The frequency response curve looks like it models the figure-8 response, due to the mild bump

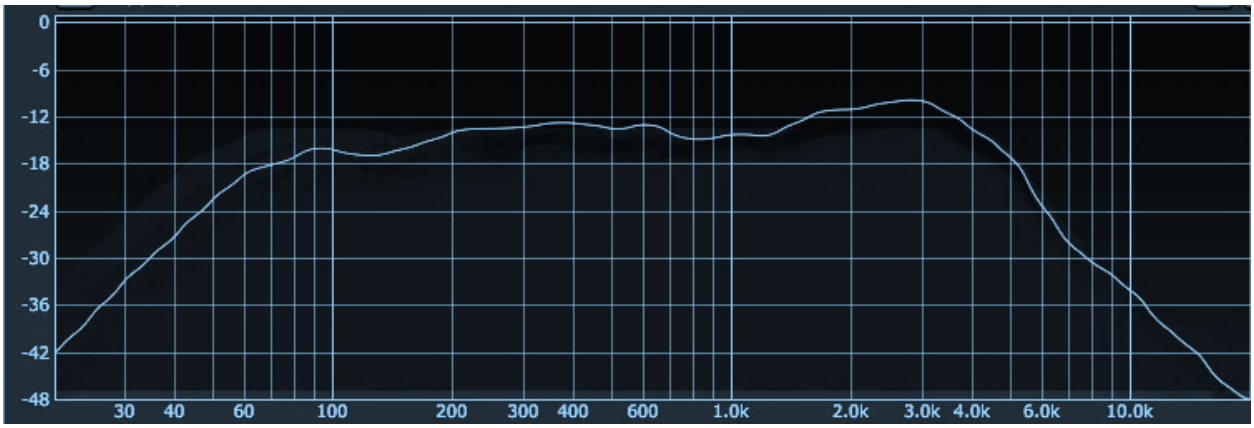
around 2 kHz, and the fast high-frequency rolloff. However, that's just a guess. Then again, what's being emulated is less important than what the mic adds to the Helix toolset.

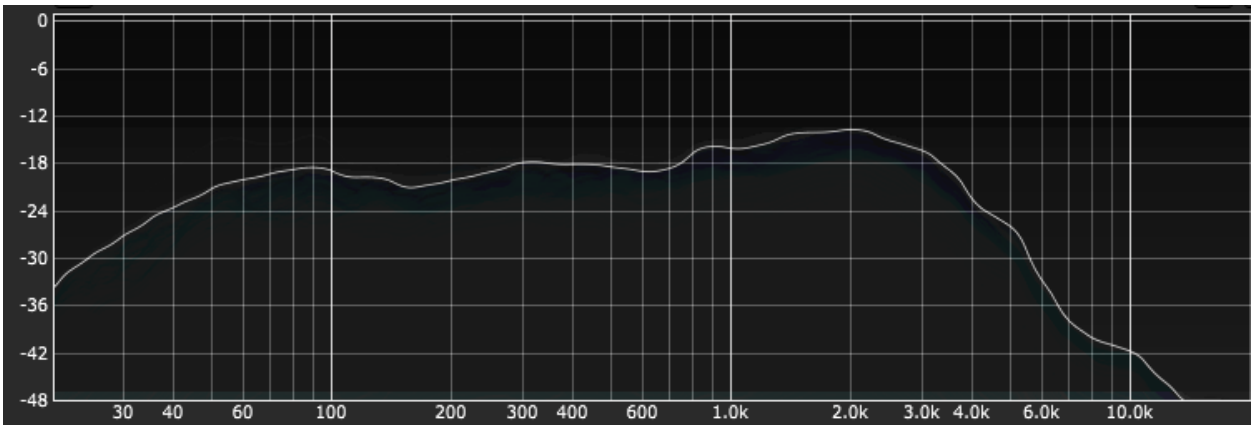
87



The 87 multi-pattern, tube, large-diaphragm condenser mic has a slight, fairly broad presence boost starting around 2 kHz. The result is a sonic crispness that works well with dry electric or acoustic guitar.

47



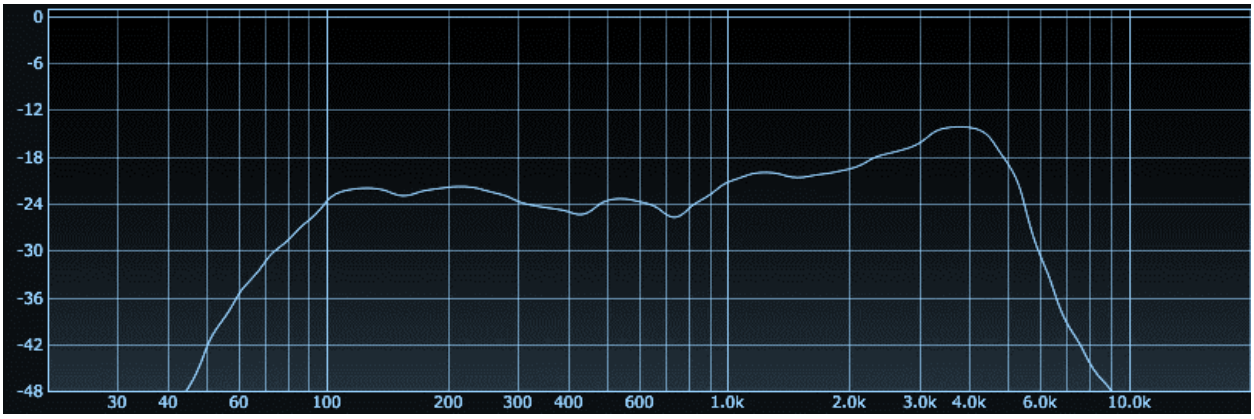


This cardioid, large-diaphragm condenser models an FET amplifier instead of a tube. Compared to the 87, it has a faster transient response and more presence.

v3.50 Bass Cabs Mics

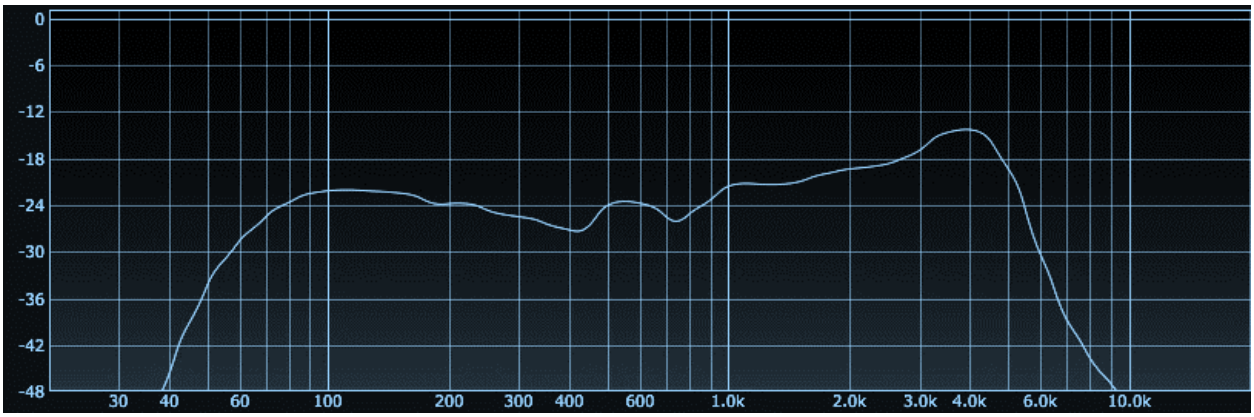
How these mics respond depends on how close they are to the speaker, because this influences the bass-boosting proximity effect. The Brute was the standard cab chosen for these measurements, so the various mic responses are influenced by the cab's response.

40 Dynamic



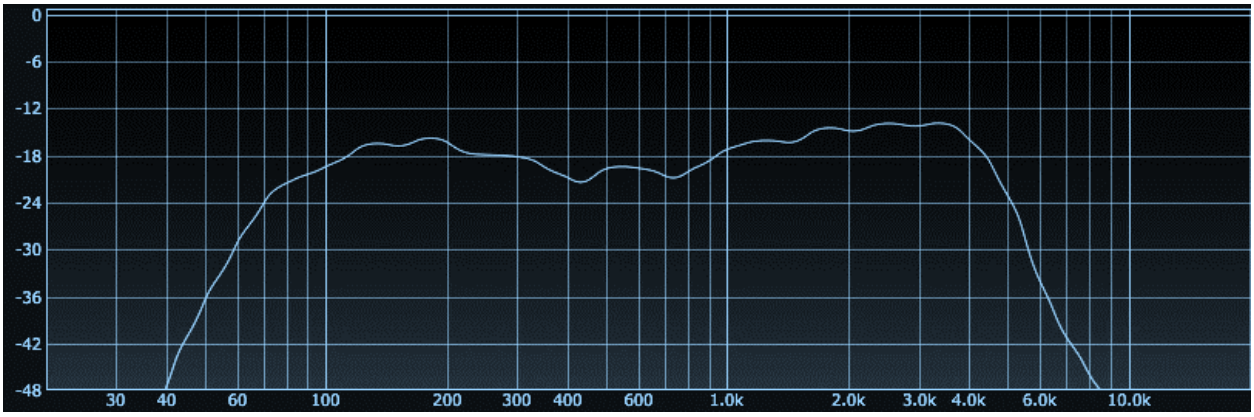
Although the PR 40 is often used with bass, it's a versatile microphone that's used in many recording scenarios. The presence lift in the upper midrange makes the PR 40 popular for vocals, as well as miking acoustic sources that need a solid bass response.

52 Dynamic



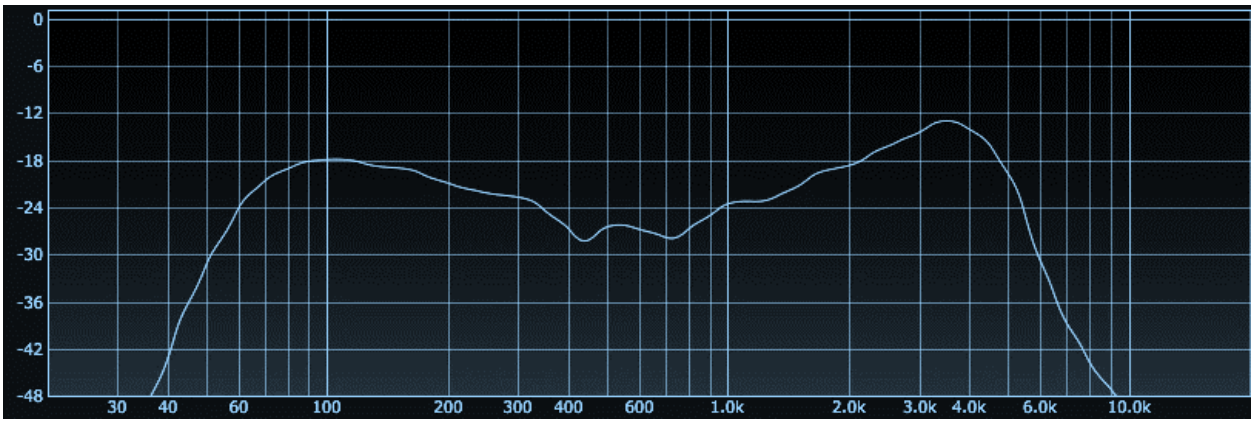
The Shure Beta 52A is designed specifically for kick and bass. For emphasizing bass with a sharp attack, Shure recommends placing the mic 1" from the speaker, on-axis with the speaker cone's center. Placing the mic further back (but still on-axis) softens the attack and reduces the bass. The 52's frequency response resembles Shure's quoted Beta 52A response with the mic placed about 2 ft. away from the speaker.

88 Dynamic



The response curve looks like what Beyerdynamic quotes for placing the mic about 4 inches away from the speaker. Closer settings increase the proximity effect, hence the bass response. The even, extended high-frequency response is another distinguishing characteristic.

D6 Dynamic



This is another mic that's clearly biased toward its intended application of miking kick drums and bass amps. The significant low-frequency boost gives a sense of fullness. The high-frequency boost provides articulation from high-frequency pick and string noises. Furthermore, the scooped mids reduce the potential for boominess and mud. This is a specialized mic that does what it's supposed to do.

Create New IRs by “Morphing” Cabs

To produce unique cab sounds, combine two different cabs in parallel, invert the phase on *one* of them, and mix them together. Each of the 65 cabs (41 Legacy cabs and 24 version 3.50 cabs) can pair with one of 64 other cabs. That gives 4,160 (65 x 64) potential combinations.

Because of the inverted polarity, mixing the cabs together cancels what the two cabs have in common. This emphasizes their differences—the various resonances and “personalities” that make each cab unique—and morphs them into a new sound.

Manage Your Expectations…

It’s impossible to predict which cabs will sound good together, and the best proportion for blending them. Many of the combinations sound dreadful, while some are novel and useful. Furthermore, a particular combination might need only some EQ, or a small change in cab parameters, for the sound to fall into place. Also, try these cabinet combinations with dry sounds. Sometimes, the results are similar to throwing a guitar’s pickup out of phase to change the sonic character.

The Setup

Fig. 6.12 shows a setup for creating IRs of morphed cabs in Helix Native.

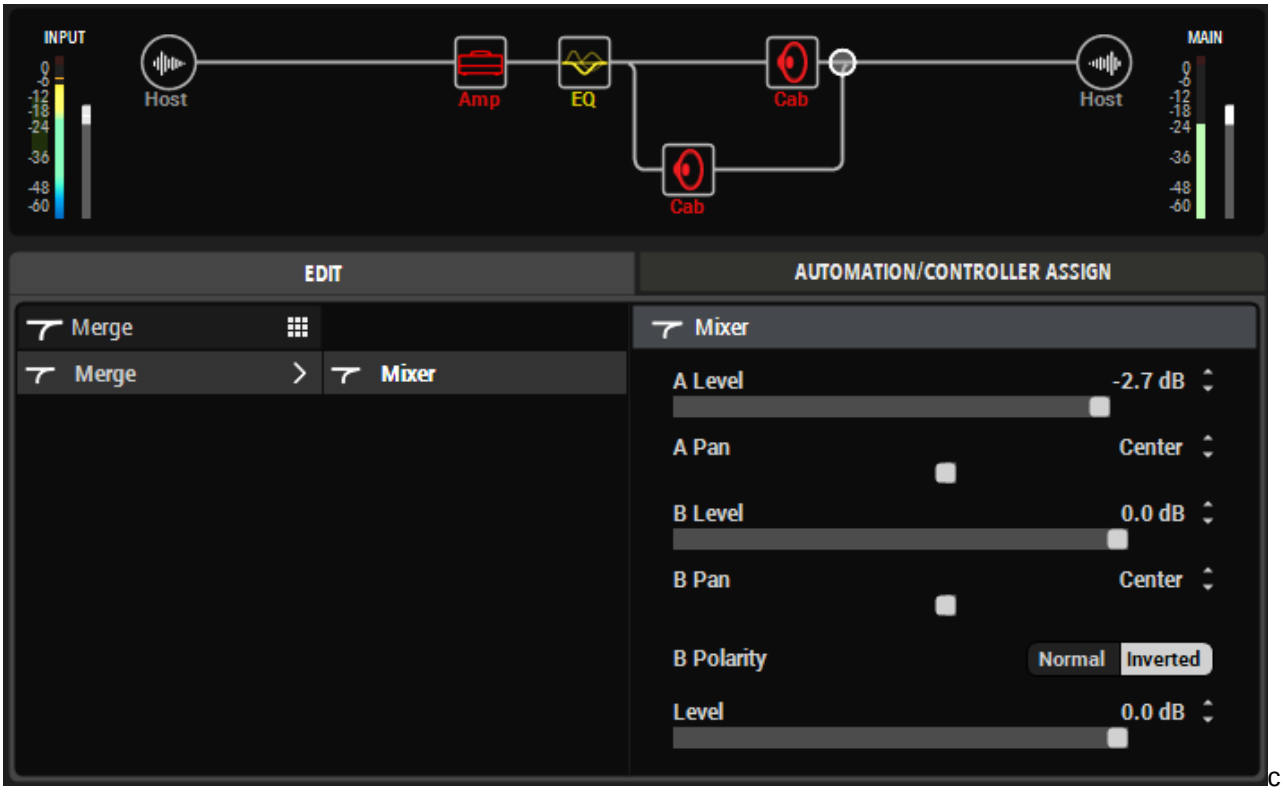


Figure 6.12 Setup to create a new, morphed cab sound prior to freezing the sound as an IR.

The heart of the process is two parallel cabs, with the Mixer inverting the B Polarity. You don’t have to create an IR to use this technique. To implement it in a preset, only the two cabs and Mixer are essential. The amp simply provides a reference as you tweak the sounds. The EQ creates optional tonal changes.

Edit the Sounds

- Initially start with equal A and B levels in the Mixer. This produces the most dramatic changes, usually with a "phasey" sound. (If you choose the same cab for both sub-paths, you won't hear anything because they'll cancel.) *The Free Files > Chapter 6 > Morphed Cabs* folder includes 16 examples of IRs obtained by morphing various cabs. They illustrate some of the more extreme settings.
- For subtle morphings, offset one cab's level by -7.0 to -15.0 dB. This adds a hint of morphing, without creating a phasey sound.
- Any changes made to a cab parameter can be dramatic, because those changes create a greater difference between the two cabs. Sometimes changing something simple, like a cab's High Cut parameter or mic position, can turn a not-so-useful sound into something unique.
- Legacy cabs in parallel with version 3.50 cabs sometimes work well together.
- Follow the mixer with a Parametric EQ to compensate for the drastic frequency response anomalies that sometimes result from summing the cabs. Even a minor boost or cut at certain frequencies can optimize the tone. In some cases, you'll need to trim the low frequencies, because that's the frequency range where many cabs differ the most.
- The cancellation caused by one cab being out of phase reduces the overall output level compared to using an individual cab. Add any needed makeup gain with one of the following:

    - The cab Level parameters
    - The Mixer's Level parameter
    - The Level parameter in post-mixer effects.

The Final Step: Create an IR

The last section in this chapter describes how to create an IR using Helix Native and a DAW. *Before creating an IR using this setup, make sure the amp is bypassed!* However, you can leave the EQ enabled if it's adding changes that you want.

Granted, the Helix cabs are fine, so this may seem like overkill. Yet this is just one more way that Helix can create sounds that aren't possible to obtain with physical amps. Some of the sounds won't be useful, but others might create surprising tones that can lead to songwriting inspiration. You'll also have unique cab sounds no one else has.

How to Create EQ-Based Cabs

Cabs are equalizers. In Helix, impulse responses and hybrid models create sophisticated frequency response curves. Before digital technology became commonplace, cabinet emulations were created with analog EQ. I started using EQ-based cabs with Helix prior to version 3.50, because EQ blocks saved DSP power compared to Legacy cabs. Replacing Legacy cabs with EQ-based cabs allowed placing DSP-hungry amps in parallel, adding more effects, taking advantage of multiband options, etc.

With version 3.50's highly efficient cabs, EQ-based cabs are less relevant—but not irrelevant. EQ-based cabs have a different sound quality compared to standard cabs or IRs. They can't replicate a specific cabinet with the same detail as an IR, but that's not the goal. Here are some reasons to emulate a cab with the Helix EQs:

- **Create a stereo Single cab.** Unlike Single cabs, stereo effects retain their stereo imaging when going through a stereo EQ-based cab.
- **Alternate sound quality.** Think of an impulse response as similar to a photograph, and analog EQ more like a painting. They both have their place.
- **New sounds.** Create a virtual cab sound that doesn't exist in the physical world. You can even create IRs from these new sounds.
- **Replace Legacy cabs with lower-power versions.** If you need to use a Legacy cab because an equivalent 3.50 version doesn't exist, an EQ-based cab can sometimes come close to the Legacy sound. Replacing the Legacy cab, even with two Parametric EQs, saves DSP power.

It's not possible for the Helix EQs to emulate the subtleties and resonances of IR- or hybrid-based cabs. However, some EQ-based cabs sound surprisingly similar. You may even prefer the EQ-based sound in some presets. Here are some tips for making EQ-based cabs:

- Guitar amps typically boost the lows and upper midrange, and scoop the lower midrange. The Parametric EQ's three parametric stages (each with frequency, boost/cut, and Q), and High and Low Cut stages, shape tones in these frequency ranges.
- Using two Parametric EQ blocks in series provides six parametric stages. This allows for more detailed frequency responses. Also, cascading the High and Low Cut filters can create steeper cutoff slopes.
- A physical cab has hundreds of subtle frequency response peaks and dips. However, some of these are more important than others for creating a particular sound. Prioritize those characteristics with the EQ.
- When it's *really* important to conserve DSP power, a single Parametric EQ stage may sound acceptable, although it won't be able to emulate specific cab characters with much detail.

You don't *need* to know how to create analog cabs. The Helix cabs are great, the new ones in version 3.50 are efficient, and there are tons of IRs available. However, learning how to create your own cabinet responses gives the background needed to customize your sound, and it can also be fun.

The *Free Files > Chapter 6 Presets > EQ-Based Cabs* folder includes 33 EQ-based cabs that use two Parametric EQs. See About the Free EQ Cab-Based Presets for more information about these free cabs. Also see Substituting EQ-Based Cabs for Stock Cabs or IRs for information on how to copy the EQ blocks, and then paste them in place of an IR-based cab. The *Free Files > Chapter 11 Favorites* folder includes 10 additional single Parametric EQ-based "eco" cabs.

So, let's open our virtual woodworking shop, and create our own cabs. However, as with the "morphed" out-of-phase cabs, manage your expectations. EQ-based cabs provide an *alternative* to IRs. They may be the only way to create a preset with the sound you want, because Legacy cabs require too much DSP power. Or, they may be

the best way to provide a particular tone. Furthermore, there's a learning curve, because setting EQ parameters is like a combination lock—all the EQ stages have to work together to create the final sound.

Test Setups for Creating EQ-Based Cabs

Choose from two main setups for designing EQ-based cabs.

Option 1 (Basic): Create Custom EQ-Based Cabs

This method works with all Helix devices that include cabs. It's the simplest option if you don't need to create variations on specific cabs, but simply want EQ-based cabs for your Helix toolchest. It's possible to create cabs with Helix hardware, but the process can be tedious. It's easier to run HX Edit or Helix Native on a computer.

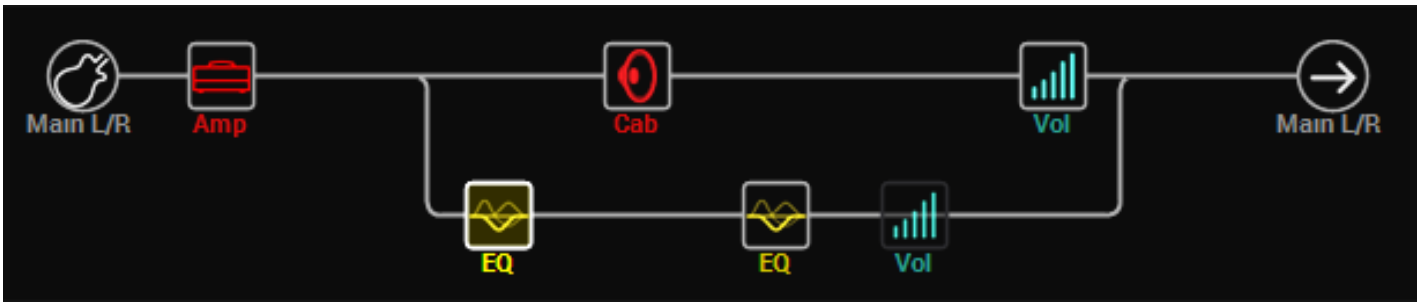


Figure 6.13 Test setup for creating EQ-based cabs.

Fig. 6.13 shows the test setup. Insert your amp of choice, then split its output. One path goes to two Parametric EQs in series. The other path goes to a typical cab. Alternating between the two paths lets you compare your EQ settings with the cab, so you don't stray too far from how cabs sound (unless you want to, of course!). To switch between the paths as you compare sounds, the most convenient methods are:

- Assign an Expression Pedal to the Splitter's Split A/B function, so you can listen to one path or the other by moving the pedal.
- Use two Volume/Gain blocks (as shown in fig. 6.13) as mute buttons so you can enable one path while muting the other one.

Play into your Helix device, and then start editing the EQs to create a cab.

Tip: With Helix Native setups in a DAW, record your playing in the DAW, loop what you played, and feed it into the test setup. Then you don't have to keep playing your guitar over and over and over and over (and over) while you A/B the EQs and cab.

Option 2 (Advanced): Create Variations on Existing Cabs (Requires Helix Native and a DAW)

Apologies in advance…we're getting into rocket science territory, so feel free to move on to About the Free EQ-Based Cab Presets. The following approach is about emulating specific cabs, like Legacy cabs, which you can then customize with the EQ's parameters.

1. Create two tracks in a DAW and insert a spectrum analyzer plug-in in each track. Fig. 6.14 shows the Spectrum Analyzer in PreSonus Studio One, but many other DAWs include dedicated spectrum analyzers. If not, some EQs (like the Waves F6) incorporate a spectrum analyzer. Or, download the free SPAN spectrum analyzer plug-in from voxengo.com.

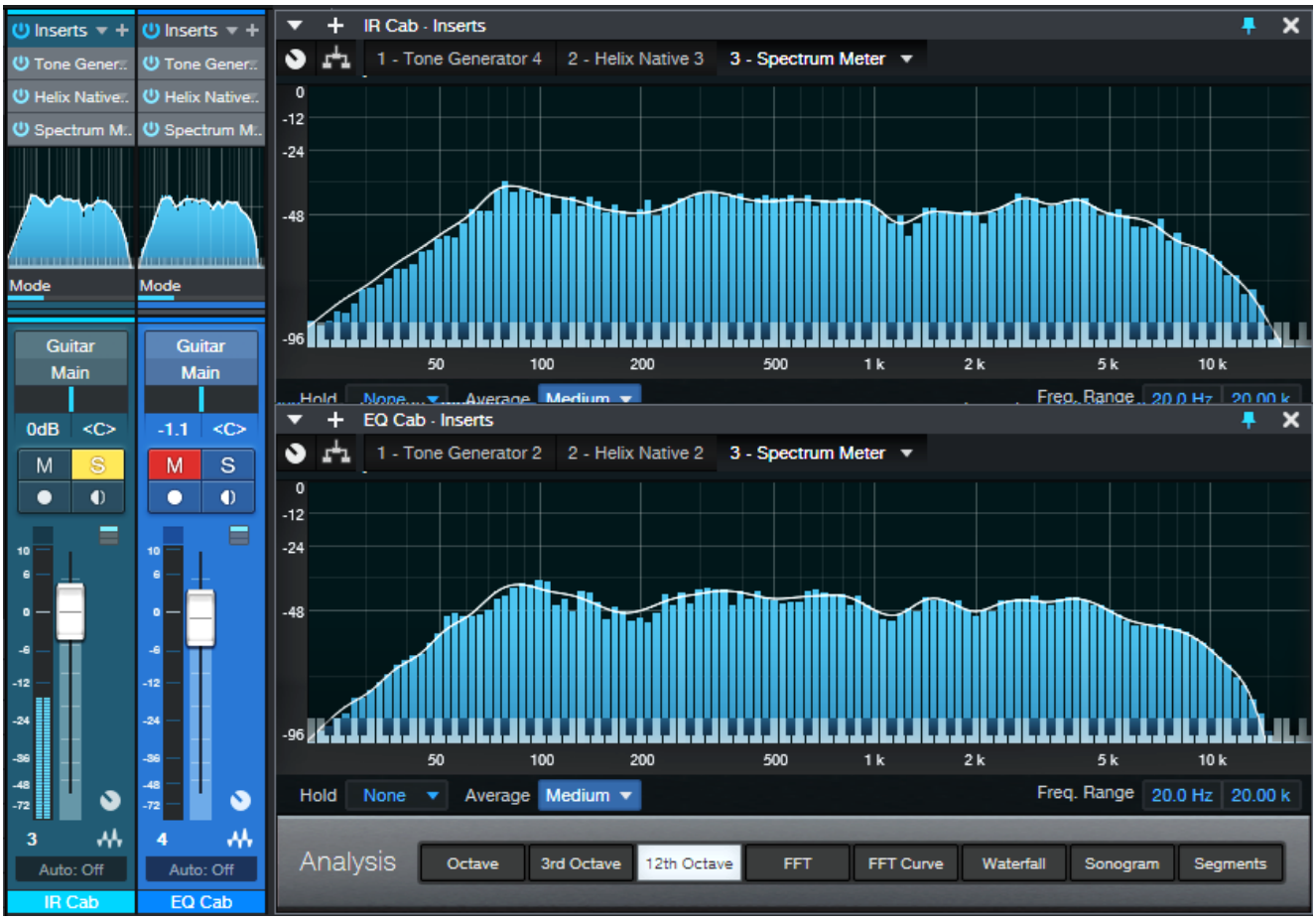


Figure 6.14 The goal is to adjust the parametric EQs so that when you switch between the source cab and EQ-based target cab, you see the same curve in the spectrum analyzers, and hear the same tonal quality in the pink noise audio. Note the similarity between the stock cab (top) and the EQ-based version (below).

2. Load a pink noise audio file into each track. Free pink noise samples are downloadable from the internet, or use a plug-in that generates pink noise. Several DAWs include pink noise generators. (Waves eMo Generator is a noise and tone generator plug-in. Virtual synthesizers may include white or pink noise.)
3. Insert an instance of Helix Native into each channel, between the pink noise generator and spectrum analyzer (see the upper left of fig. 6.14). Insert the cab you want to emulate in one Helix Native instance, and two series Parametric EQs in the other instance.
4. Follow the process described next to adjust the Parametric EQ settings.

How to Adjust the Parametric EQ Settings

The goal is editing the Parametric EQ parameters to create a response curve for either a real or imaginary cab sound. With Option 1 described above, go by your ears until you like the cab sound. Option 2 is more demanding because it's about emulating an existing cab, prior to creating your own variations. Aim for the closest match between the spectrum analyzer curves for the cab being emulated, and the curve created by editing the Parametric EQs. As you edit the EQ settings, the pink noise's curve changes. Add peaks and dips where the cab response has peaks and dips.

The parameters interact, so this process has a learning curve. For example, when you edit the Low Cut filter cutoff, the slope may not be steep enough. Editing the Low Cut filter in the second parametric creates a steeper slope, by cascading the two filters. But now, you might need to boost part of the bass range with a parametric stage, because the steeper slope attenuated some frequencies you wanted to leave in.

Conversely, cutting a range of frequencies has the same result as emphasizing the frequencies on either side of the cut. However, cuts or boosts may need to have a higher Q than what's available in the Parametric EQ. Or, they may not cut or boost enough. You then need to decide which parameters to trade off.

In fig. 6.15, the top image shows pink noise running through the 1x8 Small Tweed Cab. The middle image uses two Parametric EQ stages to emulate the frequency response curve. The colored squares correlate to the Parametric EQ parameters in fig. 6.16. Although the stock cab has more nuances, the EQ still incorporates the cab's essence. The image at the bottom of fig. 6.15 uses one stage of Parametric EQ. It's much less detailed, but still can be a viable custom cab sound.

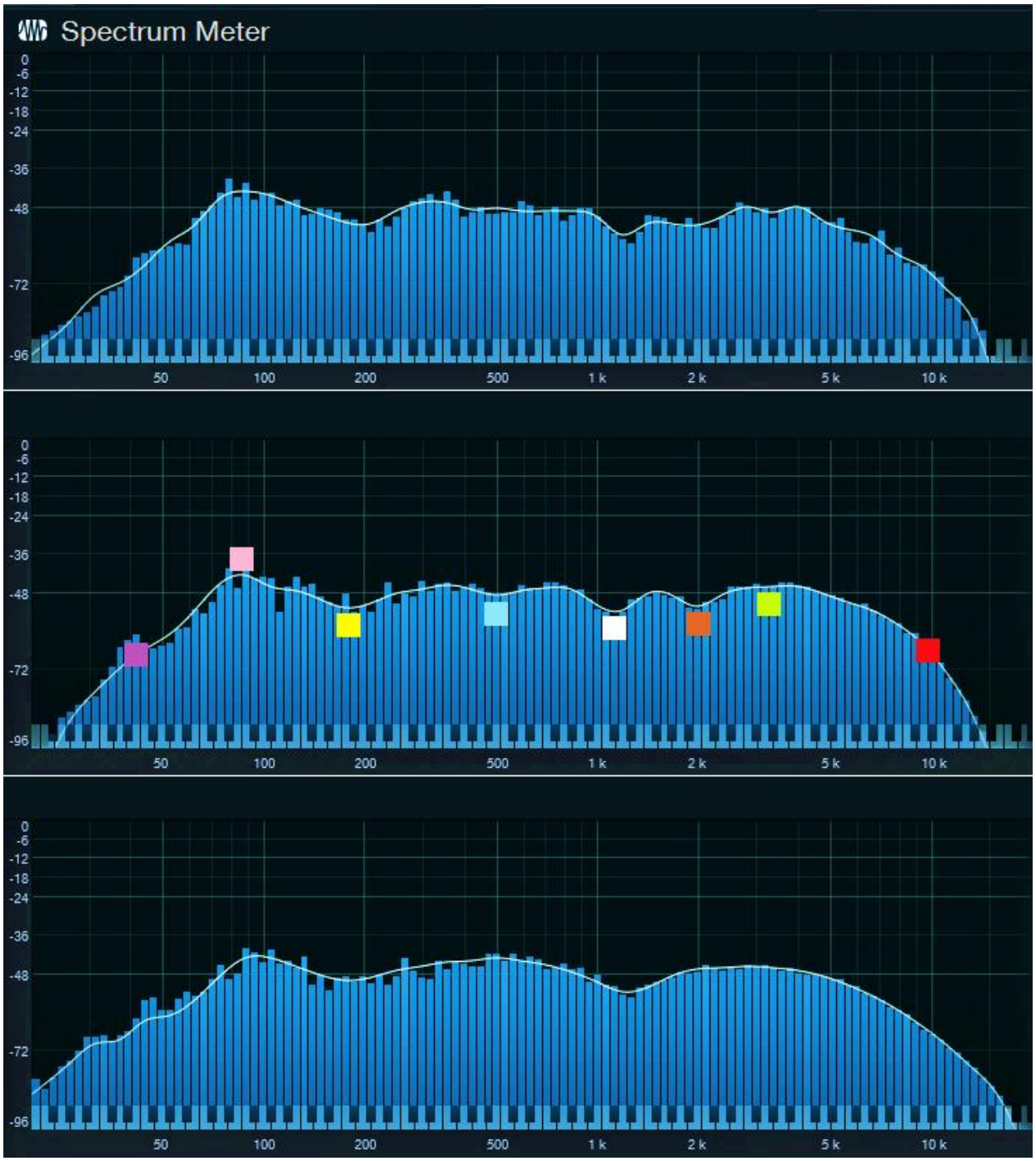


Figure 6.15 Top: The original cab response. Middle: The curve generated by editing two Parametric EQ blocks. Although the stock cab shows more nuances, the EQ still incorporates the cab's essence. The colored squares correlate to the Parametric EQ parameters in fig. 6.16. Bottom: The curve created by a single Parametric EQ block.

Fig. 6.16 color-codes the Parametric EQ parameters, to highlight what they contribute to the middle frequency response image in fig. 6.15. I usually start by adjusting the Low Cut and High Cut filters (their sliders below are colored purple and red, respectively) on both EQs to match the cab's high- and low-frequency rolloffs.

Figure 6.16 These EQ settings produce a cab sound very much like the 1x8 Small Tweed Cab.

Often, a cab's distinctive sound comes from the notches that de-emphasize certain frequencies. In this case, aside from the low-frequency boost (pink), the other stages use cuts to carve away at the response. But this isn't a given. Usually, a combination of boosts and cuts gives the best results.

After approximating the sound as closely as possible with pink noise, take off your lab coat, grab a guitar, turn off the pink noise, replace the test cab with an Amp+Cab, and insert the same amp before the two Parametric EQs. Compare the two sounds, and tweak as needed to create the variation you want.

Tip: Save multiple versions as you develop an EQ-based cab. If you become "lost in the edits," it helps to return to a previous version. Some of the previous versions may create valid sounds as well.

New, and Sometimes Improved, Cab Sounds

Although the Helix Parametric EQ doesn't have enough stages to match the digital precision of IR-based cabs, coming close may result in a different cabinet sound that's equally useful—or maybe even more so. When I replaced the Legacy 4x12 Greenback 25 Cab with an EQ-based cab, it not only saved processing power, but

sometimes, I preferred the EQ-based cab's sound (fig. 6.17). Even with the new version 3.50 4x12 Greenback 25 Cab, I still use the EQ cab sometimes with the Brit Trem Nrm amp.

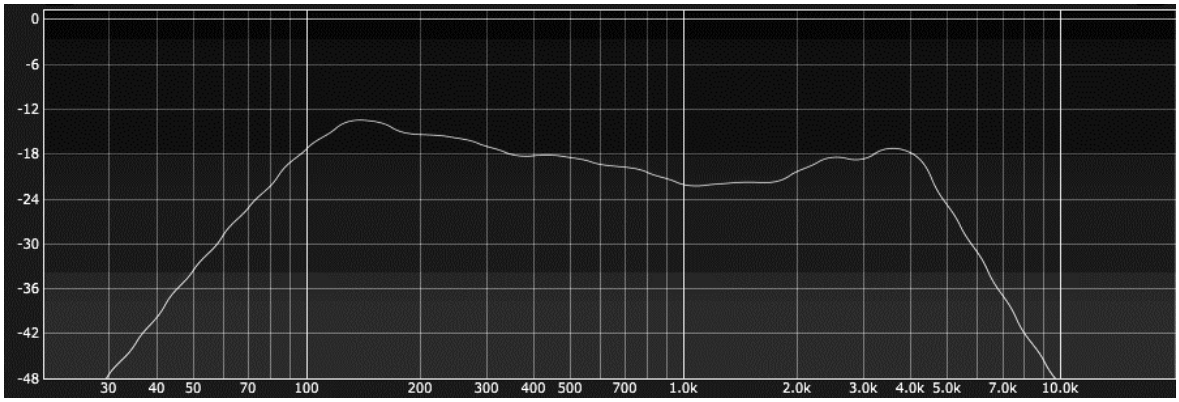


Figure 6.17 Compare this EQ cab's response to version 3.50 of the 4x12 Greenback 25 Cab. The EQ-based version tightens up the low end a bit for a more open-back cabinet sound, brings up the high end around 4 kHz to increase brightness, and has a shallower scoop between 200 Hz and 2 kHz.

Even small EQ parameter tweaks can have a *major* effect on overdriven or distorted guitar sounds. Although emulating an existing cab is a good way to start experimenting with EQ-based cabs, you can take this further if you want to create unique, custom cab sounds using EQ.

Combining Stock and EQ-Based Cabs

Helix has three potential cab types (stock, IRs, and EQ). Sometimes, combining them can give the best results.

For example, with a stereo preset, choose a stock cab for one channel, and an EQ-based, "analog" cab emulation for the other channel. There may be enough differences to give a satisfying stereo image, yet the combined sound still sounds like a stock cab. I also mix analog and stock cabs with multiband presets—some bands use stock cabs or IRs, while others use analog cabs. Mixing cab types can give a preset more color than using a single type.

About the Free EQ-Based Cab Presets

The presets in the *Free Files > Chapter 6 Presets > EQ-Based Cabs* folder include an amp (or two amps for presets with stereo Parametric EQs) and the Parametric EQs. The placeholder amp/amps give an initial impression of the EQ-based cab sound. Then, edit the preset with your amp(s) and effects of choice.

33 of the EQ-based cabs emulate selected Legacy cabs with their default settings. Some of the emulations come reasonably close, like the 4x10 Tweed P10R, 1x10 US Princess, 2x12 Double C12N, 1x12 US Deluxe, 1x12 Celeste 12H, 4x12 WhoWatt 100, 4x12 Greenback 25, 1x12 Grammatico Nrm, and Soup Pro Ellipse. Others are similar but lack detail. And some aren't close at all, like the 2x12 Blue Bell. I included those anyway, because they have their own distinctive sound.

The left-most column shows EQ versions of Legacy cabs that don't have new, 3.50 versions. The EQ versions use stereo Parametric EQs. With parallel amps, the stereo cabs can take advantage of the Mixer's Panning and Level parameters. This configuration also works well for bi-amping. Configure the Split block before the parallel path as a Split Crossover.

The middle column shows EQ cabs that have version 3.50 equivalents. These do not use stereo EQs.

The right-most column shows 10 CA-X "Eco" presets that use only one Parametric EQ stage. Before the version 3.50 cabs, these saved huge amounts of DSP power. But for saving power, they no longer offer any advantage compared to using the version 3.50 cabs. I almost decided to remove them for this book update, but they still have their uses. For example, I sometimes use them for shaping post-Distortion block tones. So, they may not be "favorites' anymore, but I left them in the *Free Files* > *Favorites* folder (see Chapter 11 on Favorites for more information).

Not Available as 3.50 Cab	Available as 3.50 Cab	Single EQ Cabs
Soup Pro Ellipse.hlx	4x12 Greenback25.hlx	CA-X Eco 01.hlx
1x8 Small Tweed.hlx	4x12 Mandarin EM.hlx	CA-X Eco 02.hlx
1x12 Celeste 12H.hlx	1x10 US Princess.hlx	CA-X Eco 03.hlx
1x12 Fullerton.hlx	1x12 Grammatico.hlx	CA-X Eco 04.hlx
4x12 WhoWatt 100.hlx	2x12 Double C12N.hlx	CA-X Eco 05.hlx
1x12 Lead 80.hlx	2x12 Jazz Rivet	CA-X Eco 06.hlx
4x12 SoloLeadEM.hlx	1x12 US Deluxe.hlx	CA-X Eco 07.hlx
2x12 Match H30.hlx	2x12 Blue Bell.hlx	CA-X Eco 08.hlx
2x12 Interstate.hlx	4x10 Tweed P10R.hlx	CA-X Eco 09.hlx
1x12 Blue Bell.hlx	4x12 1960A T75.hlx	CA-X Eco 10.hlx
1x12 Field Coil.hlx	4x12 Blackback30.hlx	
4x12 Greenback20.hlx	4x12 Cali V30.hlx	
2x12 Silver Bell.hlx		
1x12 Cali IV.hlx		
2x12 Match G25.hlx		
1x12 Del Sol.hlx		
1x18 Del Sol.hlx		
1x18 Woody Blue		
4x10 Ampeg HLF		
6x10 Cali Power		

Substituting EQ-Based Cabs for Legacy Cabs or IRs

The following scenario takes advantage of EQ-based cabs. Suppose you have presets with Legacy cabs, and there aren't equivalent 3.50 versions. However, you've been spoiled by the new cabs drawing so much less power. Although two stages of stereo parametric EQ use more DSP power than version 3.50 cabs, they draw less than Legacy cabs.

The free EQ-based cabs are embedded in 2-path presets. The easiest way to substitute EQ-based cabs for a preset's stock cabs is with two Helix-family devices, either hardware or software:

1. Open the preset with the EQ-based cab in one Helix device.
2. Open the preset you want to modify in the other Helix device. What do to next depends on your Helix-family devices:

- HX Edit can have open windows for two hardware devices at the same time. Drag-and-drop or cut-and-paste blocks between the two.
- With two instances of Helix Native, or one instance of Helix Native and a hardware device open in HX Edit, cut-and-paste blocks from one preset to the other. You cannot drag-and-drop blocks when using Helix Native if the other device is Helix hardware or HX Edit.

The process is more tedious, but doable, without two devices:

1. Load the preset that contains the EQ-based cab. Copy a Parametric Block.
2. Open the preset with the cab you want to replace. Paste the Parametric Block in its place. Save the preset.
3. If there are two parametric blocks, repeat step 1 but copy the second Parametric block. In step 2, paste the second Parametric after the first one.

Convert Mono Parametric EQ Blocks to Stereo Parametric EQ Blocks

Most of the free, EQ-emulated cabs are stereo and use stereo Parametric EQ blocks. However, some are mono. If parallel amps feed the Mixer, using stereo Parametric EQ blocks can take advantage of the Mixer's stereo panning capabilities.

There's no easy way to convert mono Parametric EQ blocks to stereo and retain the same settings. You have two options:

- Write down the block settings, change the block to stereo, and reproduce the settings you wrote down.
- If using HX Edit or Helix Native, take a screenshot of the mono block settings so you don't have to write them down. Then, change the block to stereo, and reproduce the settings from the screenshot.
- Open two instances of Helix devices (e.g., two HX Edit windows, two instances of Helix Native, or an HX Edit window and Helix Native) and edit the stereo EQ's settings to match the mono EQ.

EQ Cab Applications

I'm a fan of the WhoWatt 100 Amp, but there's no version 3.50 of the Legacy 4x12 WhoWatt 100 Cab. So, the **2B_WhoWatt100.hlx** preset (fig. 6.18) in the *Free Files > Chapter 6 Presets* folder uses an EQ-based stereo cab and the input Splitter's crossover to bi-amp two WhoWatt 100 Amps. This preset gives power chords a big, bassy bottom, and a bright presence.

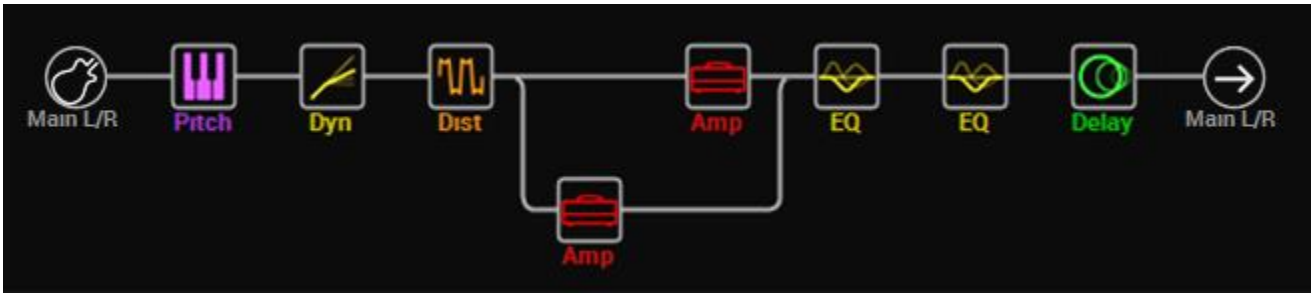


Figure 6.18 This HX Stomp-compatible preset gives a big, defined sound.

Sharing EQ-based cabs saves enough DSP power to allow for a Pitch Wham, Deluxe Comp, and Teemah! distortion before the amps. There's even enough DSP left over for a stereo Simple Delay at the output.

Creating a preset that mixes bi-amped WhoWatt 100 amps into a Legacy 4x12 Who Watt 100 cab would require several compromises compared to use EQ-based cabs:

- The Legacy cab would sum the amps into mono, so you couldn't pan them in stereo.
- If you retain the Deluxe Comp and stereo Simple Delay, you can choose from only 6 distortions (which doesn't include the Teemah!). With the EQ-based cab version, you can choose from 33 distortions.

- If you choose the Top Secret OD distortion, you can't add any more effects. With the EQ-based cab version, choosing the Top Secret OD distortion allows adding another stereo output effect (half the Modulation effects, most of the Delay effects, or three of the four Filter effects), *or* more mono effects at the input (e.g., Pitch Wham).

The bottom line is that the EQ-based cab preset sounds like summing the two amps into a Legacy 4x12 WhoWatt, except that you have a true stereo effect, and can choose from a much wider variety of effects.

Here's another example. Fig. 6.19 shows a preset with a Brit Trem Jump and 4x12 Greenback 25 EQ-based cab, preceded by the Poly Wham. With the Legacy 4x12 Greenback 25, there wouldn't be enough DSP left over to insert the Poly Wham.

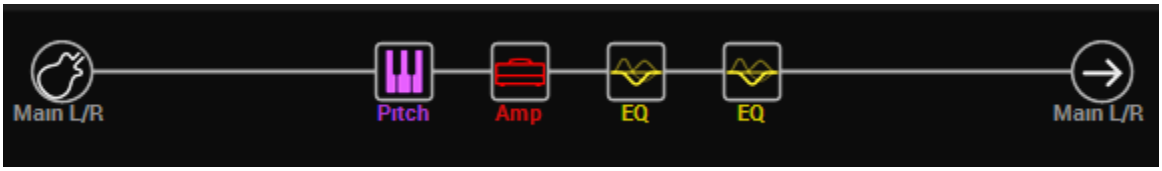


Figure 6.19 Inserting the Pitch Wham is not possible with a Legacy 4x12 Greenback 25 cab.

Fig. 6.20 shows another reason to consider EQ-based cabs.

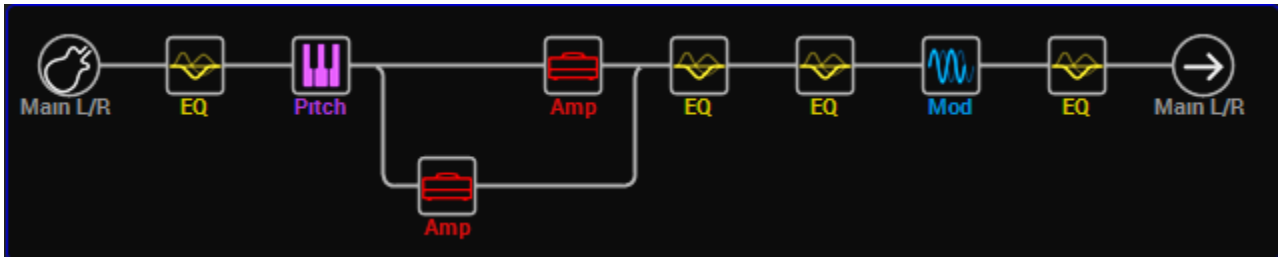


Figure 6.20 This preset is optimized for low tunings. It uses EQ-based cabs to give the desired sound, but this also frees up DSP power compared to using a Legacy cab.

The preset **2B_SonOfAPitch.hlx** in the *Free Files* > *Chapter 6 Presets* folder is based around bi-amped US Small Tweed amps, which paired with a 1x8 Small Tweed Cab prior to version 3.50. There's no 3.50 version of the 1x8 Small Tweed Cab, so the amp now pairs with a 1x10 US Princess. However, this doesn't give the same sound as the original. The EQ-based cab sounds a little closer.

Furthermore, not using the Legacy cab frees up a lot of DSP. There's a Low and High Cut filter before the Poly Capo to condition the sound for better tracking, a bi-amp split for great stereo, 4-Voice Chorus, and a Low/High Shelf EQ at the output to add some brightness to the low-tuned sound. Or, use a stereo Simple Delay (or various other effects) in place of the 4-Voice Chorus.

Finally, using EQ-based cabs can be a matter of taste. For a 3-band preset using the Jazz Rivet Legacy Amp+Cab, one path (the low band) was a 3-Band Compressed-based frequency split with an Amp+Cab and 70s Chorus. (See Chapter 9 for more information on 3-Band multiband processing.) The other path split into high and mid frequency bands, which fed stereo 70s Chorus, Double Tank Reverb, and Dual Delay.

The stereo effects were essential for a luscious, spacious sound, but they didn't leave enough DSP power for two Legacy Amp+Cabs. So, creating the sound I wanted required a DAW setup with one instance of Helix Native for the path with the two splits and Legacy Amp+Cabs. That fed a second instance for the path's effects.

For a “live” version, I used EQ-based cabs as substitutes (fig. 6.21) for the high and mid frequencies. That gave the desired sound and fit into a floor Helix. When version 3.50 came out, I replaced the EQ-based cabs with the new 3.50 2x12 Jazz Rivet Cab. However, the new cab differs from the Legacy version. If it ain’t broke…so I kept the preset as is. It’s in the *Free Files > Chapter 6 Presets* folder as **3B.Jazz Rivet.hlx**. Note that this preset works only with 4-Path Helix devices.

This story also has a happy ending with version 3.50, because I made a preset using the new 2x12 Mandarin 30 Cabs in place of the EQ-based cabs. This created a useful variation on the same essential sound. As a final note, two of the 3-Band Comps add some compression, along with serving as frequency splitters.

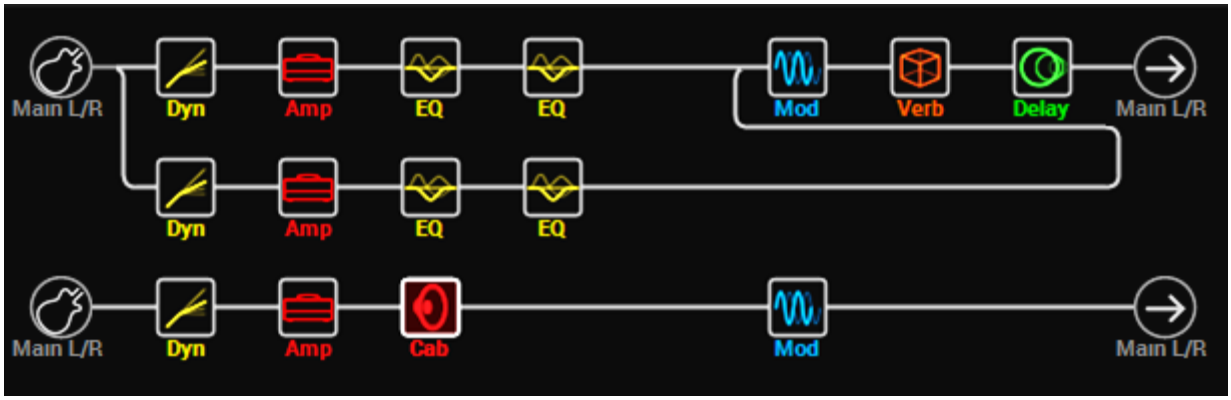


Figure 6.21 Top to bottom, the paths are high, mid, and low frequencies. The Reverb and Delay process only the highs and mids. There’s not enough DSP to replace the amp and EQ-based cabs blocks with Amp+Legacy Cab blocks.

Of course, the stock cabs are great, and the version 3.50 ones are even better. They do what they’re supposed to do—give a detailed, realistic emulation of a specific physical cab. However, don’t overlook the potential of “analog”-sounding, EQ-based cabs.

Create Cab Impulse Responses from Other Sim Cabs

If a different amp sim has a cab sound you'd *love* to use with Helix, it's not only possible—it's easy. The following process is much simpler than the challenge of trying to create impulse responses from physical amps and mics. The same process also works for capturing impulse responses from EQ-based cabs (even ones with multiple stages of EQ), morphed cabs, or capturing the sound of Helix Legacy cabs as IRs.

This process requires a DAW, and sending an *impulse* (a short audio file with a one-sample spike) into the virtual cab you want to capture. The *Free Files > Chapter 6 Presets > Helix Impulse Creation* folder includes a set of 1024 and 2048 sample impulses, in mono and stereo, for 44.1, 48, and 96 kHz projects (fig. 6.22).

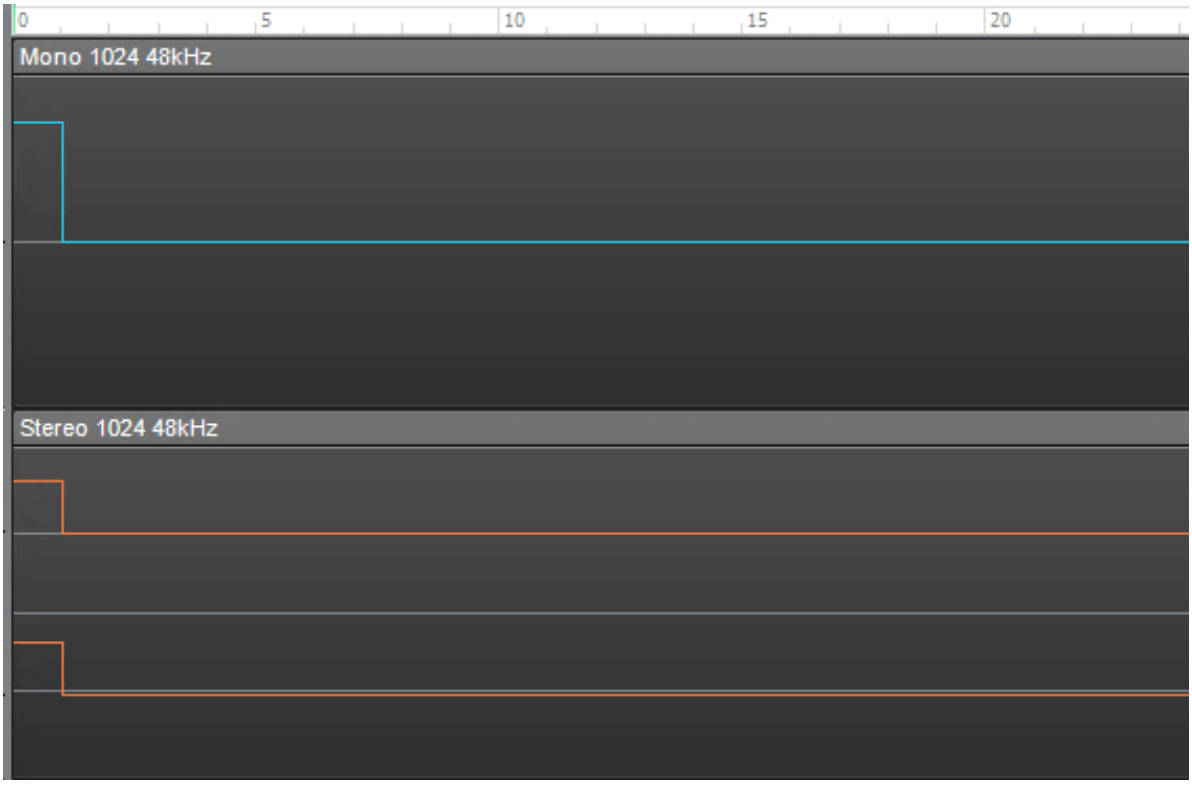


Figure 6.22 The impulse is simply a 1-sample spike. The calibration at the top is in samples.

To start, open a new project in your DAW, and choose its sample rate. You'll create the new cab's impulse response in your DAW, and then export it as a WAV file.

I recommend using a 48 kHz sample rate for your DAW's project, and loading a 48 kHz sample rate impulse into a track. Helix automatically converts an imported WAV file to 48 kHz, 32-bit, mono, with a 2,048 sample length. So, give Helix what it wants—load the 48 kHz, mono, 2,048 sample file into your DAW. (The other files are included in case you get further into this, and work with other applications that require different impulse files).

After loading the impulse into a DAW's audio track, insert the cab whose impulse response you want to capture as an effect for that track. Bypass all amps and effects—you want *only* the cab to process the impulse.

Finally, render the track. This will meld the impulse with the cab and create the impulse response file that you load into an IR block. How to render varies with the DAW, but the general procedure is:

- Select the audio clip. Then:
    - In Pro Tools, choose the Commit function.
    - In Studio One, bounce to a new track.
    - In Ableton Live, right-click on the Clip. Select Freeze. Right-click again and select Flatten.
- After rendering, I recommend normalizing the clip to -3 dB. With Ableton Live, you can check Normalize when you export (see next).
- Export the normalized clip:
    - In Pro Tools, export the clip as a file from the Clips List.
    - In Studio One, drag the clip onto the desktop, or to a folder in the Files section of the Browser.
    - Ableton Live cannot export based on a specific sample length. so, choose the minimum length. Helix will trim it to 2048 samples when it's imported.

Now you have an impulse response you can load into an IR block. Cool!

Amp Sweetener Impulse Response

The **Amp Sweetener** impulse response in the *Free Files > Chapter 6 Presets* folder inserts after Amp+Cab blocks. It's designed to reduce high-frequency "fizz" with the following high-gain Amps+Cabs:

- Interstate Zed
- Derailed Ingrid
- Mandarin Rocker
- Moo)))n Jump
- Brit Plexi Jump
- Brit 2204
- Placater Dirty
- Cartographer
- German Mahadeva
- German Ubersonic
- Cali Rectifier
- Archetype Lead
- Solo Lead Crunch
- PV Panama
- Revv Gen Purple
- Revv Gen Red
- Das Benzin Mega
- Line 6 Elektrik
- Line 6 2204 Mod
- Line 6 Fatality

To use the Amp Sweetener impulse response with Helix:

1. In HX Edit or Helix Native, click on the IRs tab.
2. Drag the Amp Sweetener.wav file into the collection of IRs.
3. Insert an IR block after a high-gain Amp+Cab, like Mandarin Rocker.
4. Load the Amp Sweetener IR into the IR block.
5. You'll need to increase the IR block level. Start around -5 dB, and then adjust as necessary.
6. Enable/bypass the block to hear the difference.

The Amp Sweetener does alter the Amp+Cab's tone somewhat. This might not be an issue. If it is, use the Simple EQ to cut the Mid Gain a bit around 500 Hz to compensate (the frequency depends on the amp).

Chapter 7 | COMBI-BAND PROCESSING

Combi-Band processing improves on bi-amp processing and works with any Helix family device. It delivers some of the benefits of 3-Band processing with HX Stomp and HX Effects, even though they have only two paths.

Combi-Band processing delivers some of the benefits of processing three frequency bands—low, midrange, and high—with only two paths. One path carries the **midrange** (mid) frequencies. The other path carries the **low+high** frequencies.

For some sounds, this improves on sounds that are possible with bi-amping. Bi-amping splits a guitar into only two bands, at a particular crossover frequency. Finding a "sweet spot" where there's a subjectively equal balance between the high and low bands is not always easy. With a Combi-Band preset, the effects in one path handle the all-important midrange frequencies, while the effects in the other path process the low *and* highs. This can provide a more even balance than bi-amping.

This chapter shows three ways to split the input into mids and low+highs. They trade off accuracy, processing power, and tone, so you can decide what to prioritize for a preset. The three options are included as Combi-Band templates in the *Free Files > Chapter 7 Presets* folder. The folder also includes several finished Combi-Band presets you can load for instant gratification, without having to know how the process works. Combi-Band presets begin with the prefix **CB**.

Template 1: Create Combi-Band Paths with the 3-Band Comp

Using two 3-Band Comps provides the most accurate mid and low+high path separation but requires the most DSP power. Fig. 7.1 shows the start of a Combi-Band preset, which uses the 3-Band Comp as a crossover. This resembles the 3-Band Comp technique [for bi-amping with independent paths.]

Fig. 7.1 How to split signals for Combi-Band processing. The input Split is a basic Y-Split.

1. Insert a 3-Band Comp into Path A. We're not using Path B yet.
2. Change the 3-Band Comp's Lo Thresh, Mid Thresh, and Hi Thresh parameters to 0.0. This defeats compression, which turns the 3-Band Comp into a three-band splitter. The Lo X Freq and Hi X Freq parameters split the audio as follows (fig. 7.2):

- Frequencies below the Lo X Freq (low band)
- Frequencies between the Lo X and Hi X Freq (mid band)
- Frequencies above the Hi X Freq (high band)

3. Start by "tuning" the midrange. Set the Lo Gain and Hi Gain controls to -60.0 dB, and the Mid Gain control to 0.0.
4. As you play guitar, adjust the Lo X Freq and Hi X Freq controls to focus on the midrange. Start with Lo X Freq = 300 and Hi X Freq = 1.5 kHz. With these settings, the midrange band covers 300 Hz to 1.5 kHz.
5. Next, tune the other bands. Set the Mid Gain control to -60.0 dB, and then increase the Lo Gain control to 0.0. Verify that you can hear the low frequencies well.
6. Set the Mid Gain and Lo Gain controls to -60.0 dB, and then increase the Hi Gain control to 0.0. Verify that you can hear the high frequencies well.

Figure 7.2 The mid and low+high path band parameters are the same, except for the ones highlighted in light blue.

7. The goal is for the 3-Band Comp's Lo, Mid, and Hi bands to cover approximately equal parts of an instrument's range. Ideally, each range should have roughly the same perceived level. If needed, tweak the Lo X Freq and Hi X Freq parameters to obtain the best balance among the three frequency ranges.
8. Now that you've "focused" the crossover frequencies to create the three bands, copy the 3-Band Comp, and create a parallel path (Path B).
9. Make sure the Input Split is set to Split Y, with both Balance parameters at Center.
10. Edit the mid path's 3-Band Comp. Set its Mid Gain to 0.0 dB, and the Lo and Hi Gain to -60.0 dB.
11. Edit the low+high path's 3-Band Comp. Set its Mid Gain to -60.0 dB, and both Lo and Hi Gain parameters to 0.0 dB. The two 3-Band Comp settings now look like fig. 7.2.

Tech Talk: Why the 3-Band Comp Is the Most Accurate Way to Split

The 3-Band Comp splits the bands evenly. If the Lo, Mid, and Hi Gain controls are the same, the output is a flat response. The tradeoff for this precision is that the 3-Band Comp requires more DSP power than other methods. This may limit the number of amps or effects that you can add. (Note that you can emphasize or de-emphasize one band more than another by altering its gain.)

Fig. 7.3 shows pink noise being fed through two 3-Band Comps—one for the mid band, and one for the low+high band. A spectrum analyzer displays the frequency response and levels after filtering.

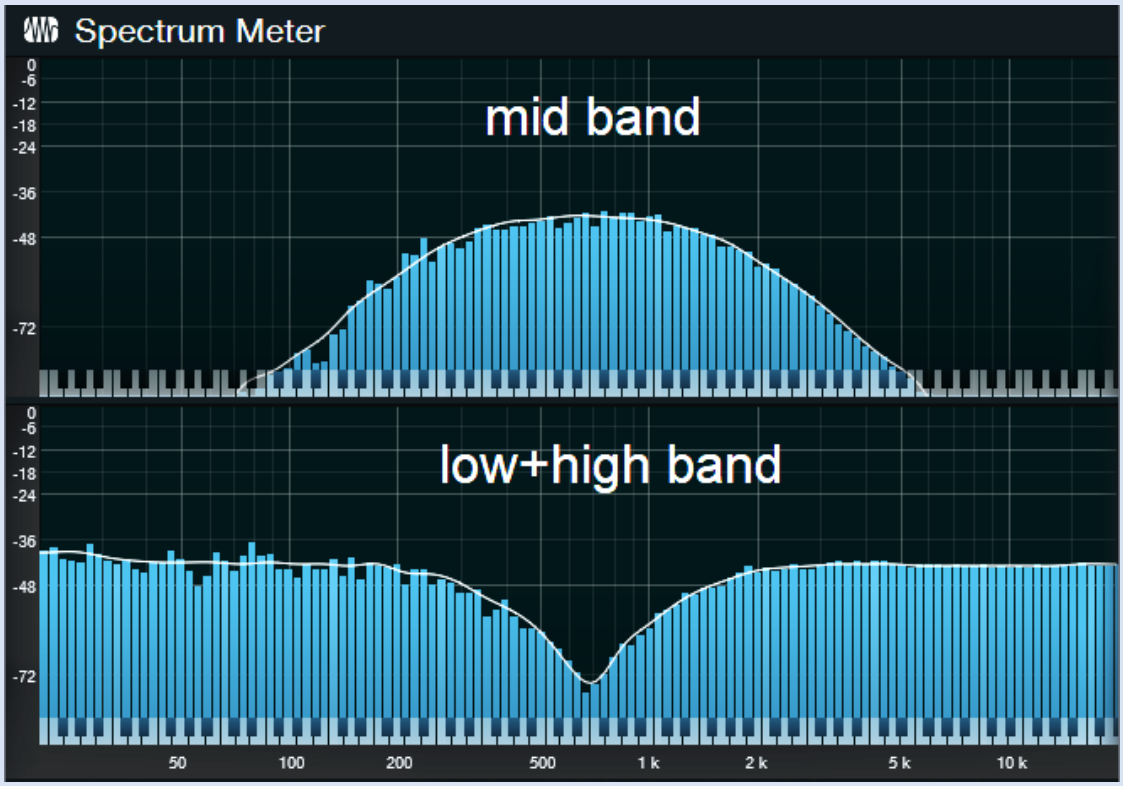


Figure 7.3 The Lo X Freq is 300 Hz, and the Hi X Freq is 1.5 kHz. This spectrum analysis shows the amplitude at different frequencies for the two bands. Adding these bands together gives an essentially flat response.

CB_Template3BDyn (fig. 7.4) is a preset template for Combi-Band processing. With stereo presets, the Mixer can pan the midrange and low+high bands.

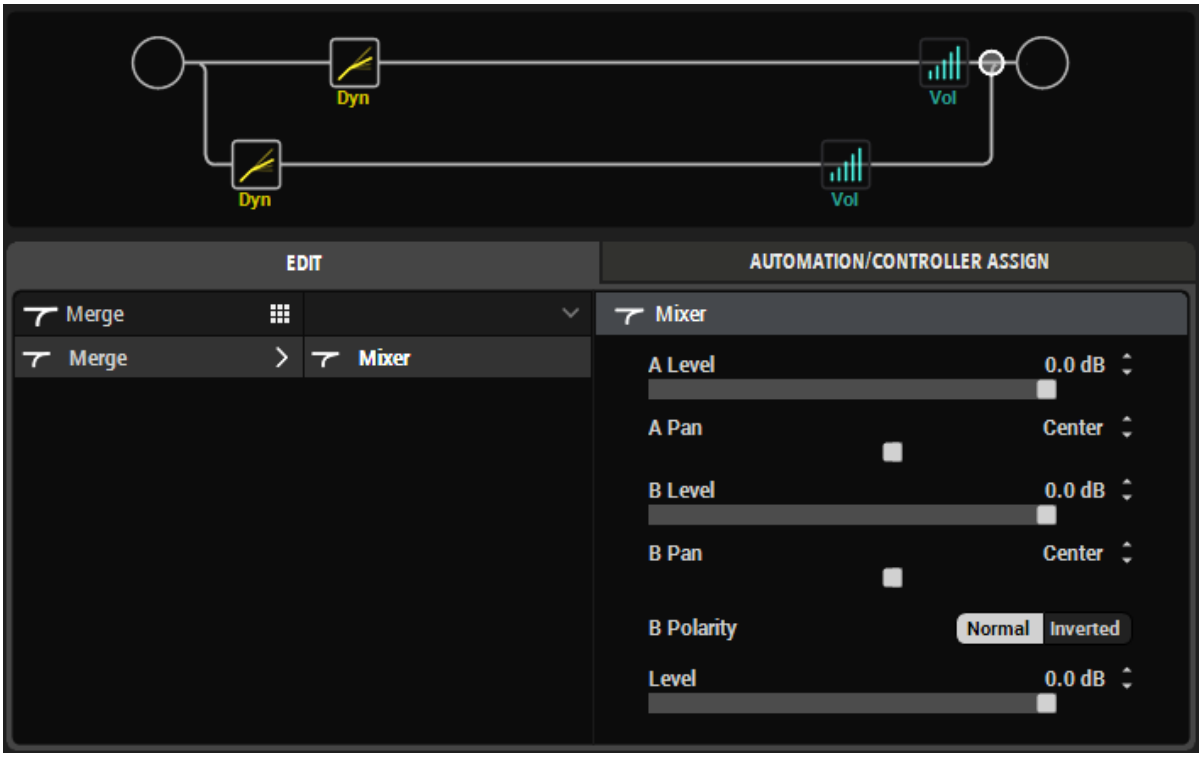


Figure 7.4 Combi-Band Template. The Vol blocks are mute buttons, for convenience when editing. After optimizing sounds in the two paths, you can remove the Vol blocks to add other effects.

The Gain Blocks

The Vol (Gain) block mute button at the end of each path can mute one path while you edit the other path's parameters. This simplifies optimizing amp and cab parameters for the frequency range they're processing. After optimizing, remove the Vol blocks to reduce processing power, or add more effects.

To create a mute button, set the Gain block's parameter to -120.0 dB. Enable the Gain block to mute the associated path. Bypass the Gain block to hear the associated path.

The Mixer

The Mixer adds considerable flexibility:

- With stereo, pan the two paths as desired.
- Choosing Inverted for the B Polarity parameter may give useful tones with mono presets.
- Adjust the balance of the two paths with the level parameters.

Tip: You can use the Mixer's level controls instead of the Gain block mute buttons to compare the two paths.

Template 2: Create Combi-Band Paths with Simple EQ

The Simple EQ requires less processing power than other EQs, but still gives good frequency-band separation. The fixed High and Low frequencies require using a default 400 Hz midrange center frequency (it's midway between the high and low bands). Fortunately, that frequency works well for many applications.

The template **CB_TemplateSimpl** (fig. 7.5) uses EQ to split the audio instead of the 3-Band Comp acting like EQ. The Mixer section is the same.

Figure 7.5 Combi-Band template using the Simple EQ.

Fig. 7.6 shows the parameter settings for the mid and low+high bands.

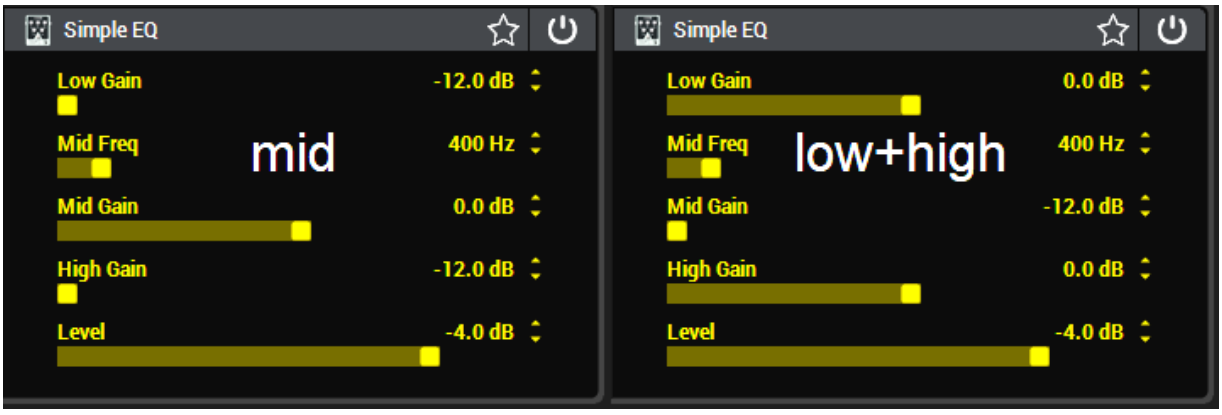


Figure 7.6 Insert the left EQ in the mid band, and the right EQ in the low+high band.

Template 3: Create Combi-Band Paths with Shelf EQ

The template **CB_TemplateShelf** uses Low/High Shelf EQs for splitting the audio. It looks the same as the template for Simple EQ-based splitting (fig. 7.5). This technique provides less separation than the previous two options because the two bands overlap more. However, for some presets this can be an advantage. For example, with distortion the sound "splits the difference" between a single-band and multiband amp. (Note: The Shelf EQ requires a little more DSP power than the Simple EQ.)

The mid path EQ (fig. 7.7, left) turns down the low and high shelves. This makes the mid frequencies more prominent. The low+high path EQ (fig. 7.7, right) turns up the high and low shelves. This boosts the low and high frequencies, so the mid frequencies are softer by comparison.

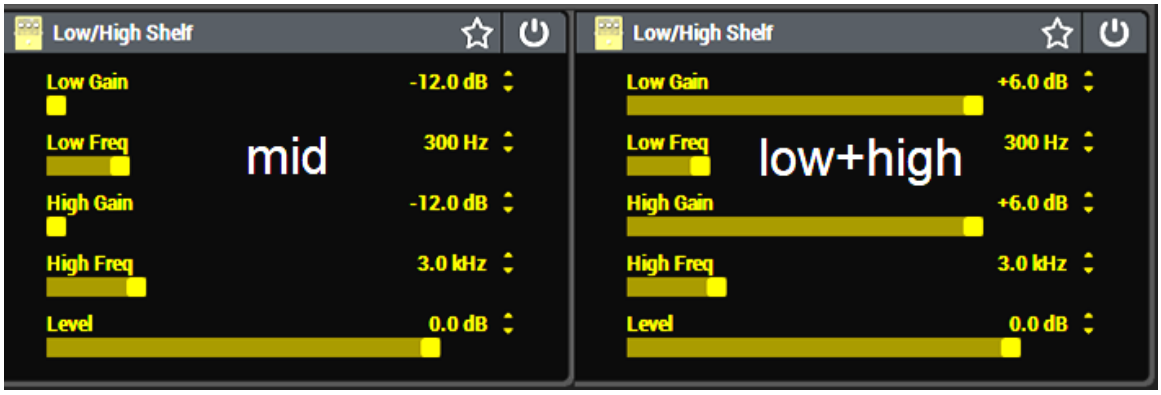


Figure 7.7 Mid band Shelf EQ parameters (left), and Low+High band parameters (right).

Comparing the Combi-Band Techniques

The *Free Files* > *Chapter 7* > *Presets* folder includes four test presets. A single-band preset, **CB_Test_1-Band**, uses default Amp+Cab parameter settings to provide a reference. The other three test presets let you try out each multiband splitting option. The test presets are not optimized for the best tone. They simply provide a way to compare the effects of the three splitting methods. The test presets are:

CB_Test_3BndComp
CB_Test_SimpleEQ
CB_Test_Shelf

Fig. 7.8 shows the **CB_Test_3BndComp** test preset. The others are the same, except the input blocks are EQs instead of dynamics. The output levels were adjusted for the same perceived level, so that the evaluation would be influenced solely by the tone. All test presets use the default **Moo)))n Brt** Amp+Cab.

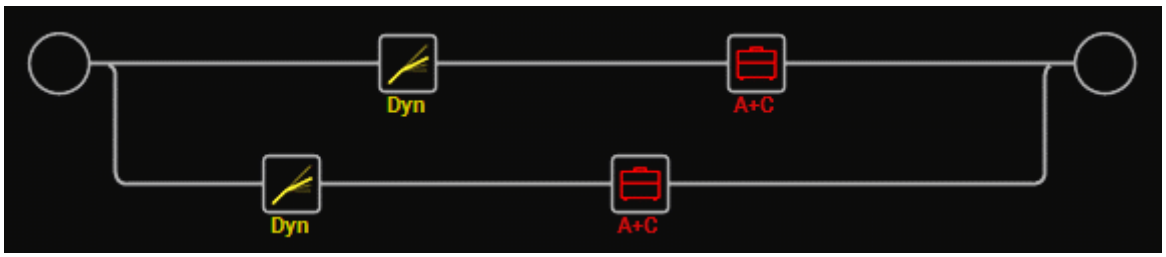


Figure 7.8 **CB_Test_3BndComp** test preset.

All three options are useful in their own way. Let's summarize.

3-Band Comp

This gives the most accurate frequency split and provides the most flexible way to split audio into three bands. The main attributes are:

- Mixing the bands together gives the flattest frequency response.
- Provides the best separation between bands.
- Requires the most DSP power of the three options.
- Less signal goes into each parallel effects chain, due to the input filtering. With distorted sounds, this is like pulling back on a drive parameter, and might sound too "clean." Turn up amp or distortion drive parameters to compensate. Or, you may prefer the more articulated sound.

Simple EQ

This is an overachiever:

- Requires the least amount of DSP power.
- There's a decent amount of separation between bands, but not as much as with the 3-Band Comp.
- Sums to a reasonably flat response.
- Distorted amps will be driven harder than with the 3-Band Comp, because the bands overlap somewhat.

Shelf EQ

This has the least amount of separation between bands, so it combines single-band and multiband elements:

- The overlap between bands feeds more audio into each band than with the other options.
- Distorted sounds are "dirtier" than the other two options (however, you can turn down drive-related parameters to clean it up).
- The Shelf EQ requires a little more processing power than the Simple EQ, but not as much as the 3-Band Comp.

Let's listen to some practical examples of Combi-Band processing.

Combi-Band Delay

Delay is well-suited to Combi-Band techniques. Any delay adds a rhythmic element, but the Combi-Band approach doubles the potential with two independent delays. This preset works well with rhythmic stabs that leave some space between upstrokes or downstrokes. Then, sustained chords or notes don't cover up the delay's tail. A reggae-style rhythm guitar is a great match for the **CB_SimpleDelay** preset (fig. 7.9).

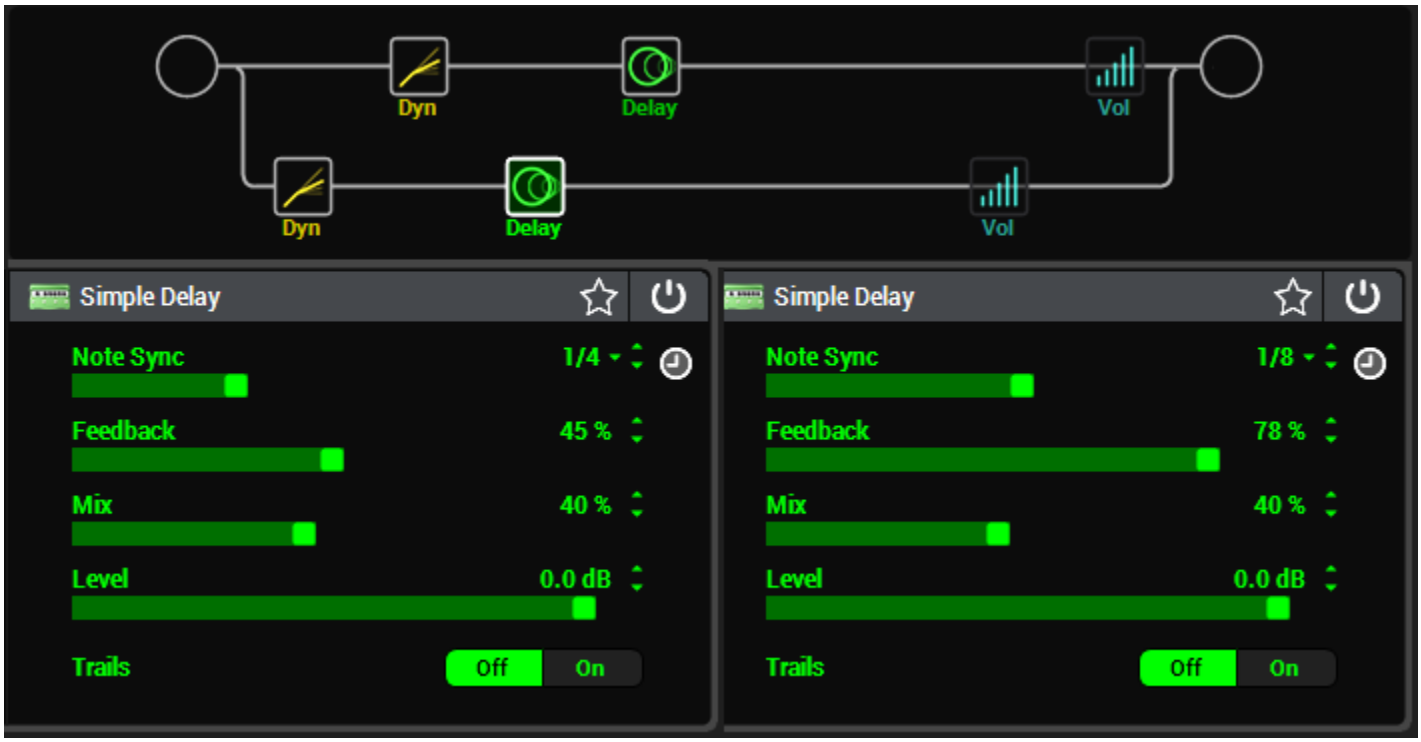


Figure 7.9 Combi-Band delay, using two Simple Delays.

3-Band Comps do the splitting. With rhythm guitar, try a longer delay for the midrange (Path A), and a faster delay on the low and high frequencies (Path B). The shorter delay requires more feedback to have the "tail" last as long as the longer delay.

If you don't use Note Sync for the delay time, consider choosing rhythmically related delay times (e.g., 125 ms and 250 ms).

Customization

- A dotted value delay, like a dotted eighth note, combined with a non-dotted rhythm (like a quarter-note) gives a feeling of motion.
- Even with a mono output, using a stereo Simple Delay gives a more rhythmic effect.
- For a triple-echo effect, use a mono Simple Delay for the midrange, and a stereo Dual Delay for the low and high frequencies (or the reverse). Try a quarter-note delay for the Simple Delay, and an eighth note and sixteenth note for the two Dual Delay times.
- Vintage Swell works well for one of the paths.
- The preset defaults to stereo but collapses perfectly to mono.
- There are a lot of options! Mute a path, then choose different delay blocks for the other path to hear how they affect the sound.

CB_SweepEcho is a fun variation on this preset. The Sweep Echo block (fig. 7.10) replaces the Simple Delay in the Low+High path. The stereo Sweep Echo gives great imaging effects, but mono works well too.

Figure 7.10 Settings for the Sweep Echo block that replaces the low+high path's Simple Delay.

Combi-Band Tremolo

CB_Tremolo (fig. 7.11) is a modern tremolo sound for EDM and other rhythmic musical applications. It uses two 3-Band Comps for frequency splitting. Each band includes one tremolo.

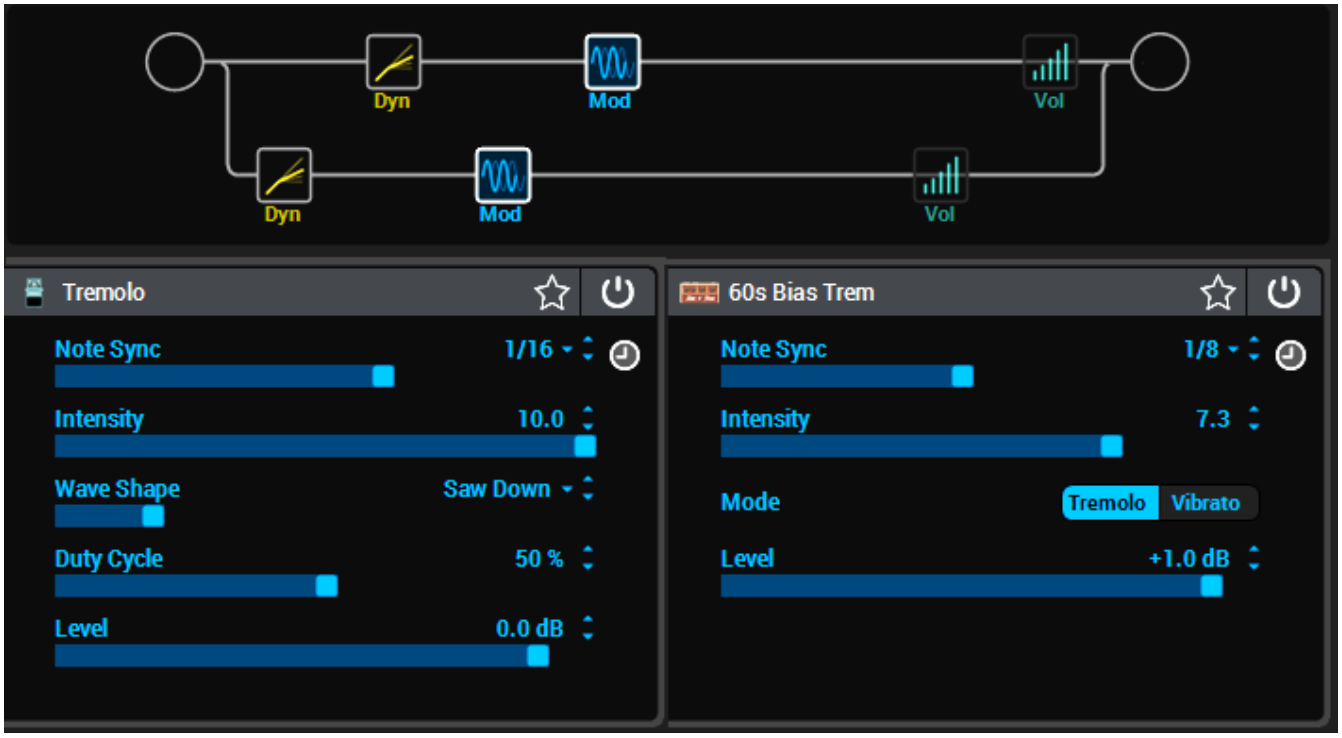


Figure 7.11 Combi-Band Tremolo preset, optimized for note sync.

Path A's Tremolo (midrange) uses a Saw Down waveform to create a series of 16th notes. The rapid decay after the initial attack provides a pulsing effect. Other waveforms work well too. Path B (low+high frequency bands) uses the 60s Bias Trem. The note sync is slower, so this path "rolls" while the midrange path pulses.

As a reminder, the Gain blocks simplify muting an individual path while you optimize the other path's settings. Remove them once they're no longer needed.

Customization

- In Path B, follow the 60s Bias Trem with another Tremolo set for Note Sync = 1/2, Intensity = 10.0, Wave Shape = Square, and Duty Cycle = 50%. Using a square wave for the Tremolo modulation source switches the 1/8th-note pattern's audio on and off every half-note. This combines a solid rhythm with the pulsating midrange Tremolo in Path A.
- Edit the two tremolos for polyrhythms. For example, set Path A's Tremolo to a dotted 1/4-note (you may want to pull back its volume a bit), and choose 1/16th notes for Path B's 60s Bias Trem.
- Another polyrhythmic trick is to insert the tremolos of your choice, but use different dotted-note values in Paths A and B. Follow each of these with Tremolos set for 1/8th sync, and sine wave modulation. This imposes a common rhythm on the two audio streams that have dotted tremolo rhythms.

Tip: To sync the tremolos while playing, briefly push the Tap Tempo button at a measure's downbeat. This resets the tremolos so that their cycles start simultaneously.

Combi-Band Phaser

Phasers have a somewhat clichéd sound, but the Combi-Band approach makes the **CB_Phaser** Preset sound more subtle. It excels in stereo yet is mono-friendly.

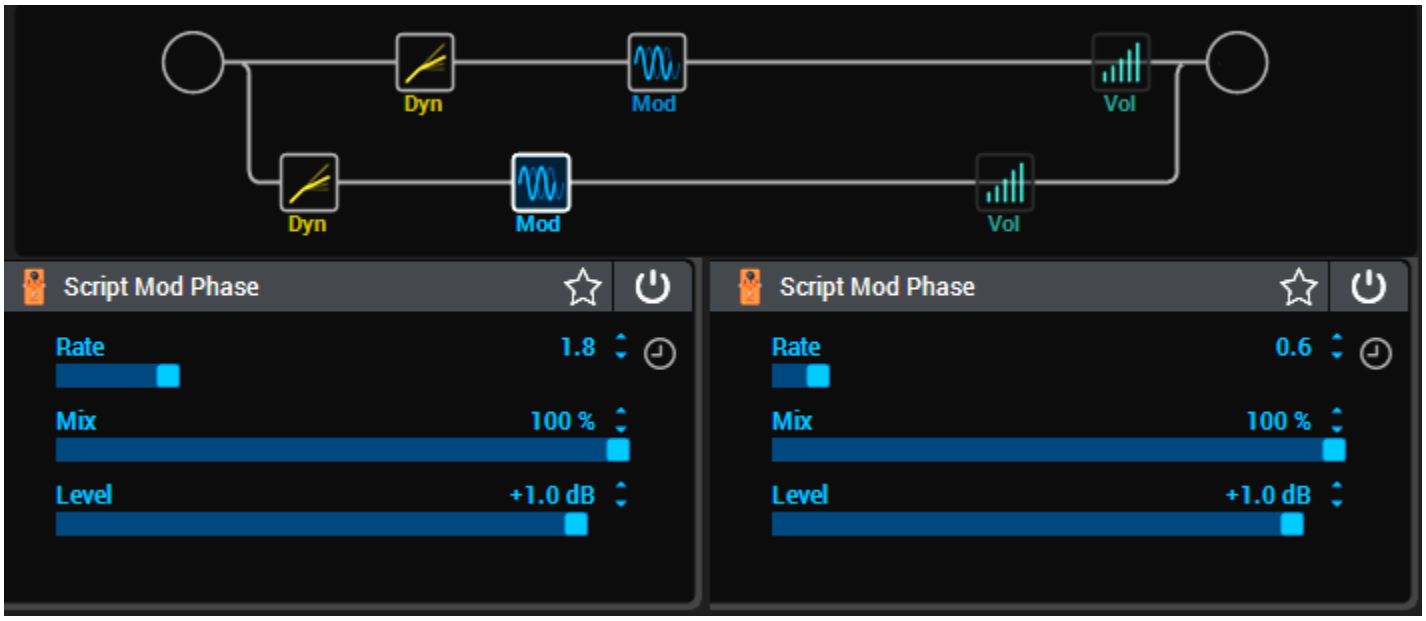


Figure 7.12 This preset gives a fresh, subtle phaser effect.

The Script Mod Phase works well for this application. The settings aren't particularly critical. Fig. 7.12 shows the midrange (Path A) filter settings on the left, and the low+high frequencies (Path B) settings on the right. A faster speed for the midrange frequency path works well.

This preset is compatible with stereo or mono. The preset in the *Free Files > Chapter 7 Presets* folder sets the pan settings to Left 70 and Right 70 (fig. 7.13). Hard left/right panning gives a less subtle stereo effect.

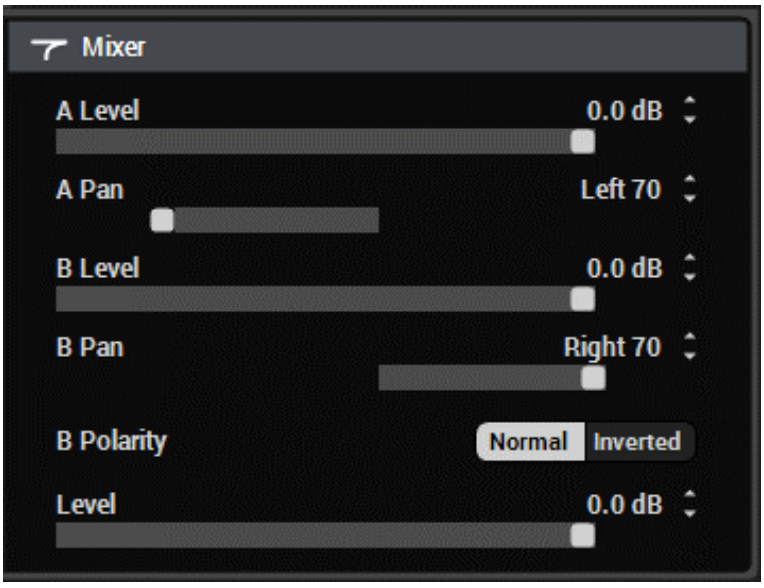


Figure 7.13 The Path A Pan leans left, while the Path B Pan leans right.

Combi-Band “Amp Replacement”

Amp replacement substitutes a better-sounding multiband amp and cab for a single-band amp and cab. HX Stomp has enough processing power to handle most Amp+Cabs in parallel, especially with version 3.50 cabs. However, Combi-Band operation requires a block before each Amp+Cab to split the input into different bands. With some combinations of amps, cabs, and effects, adding these may use up the available DSP power.

Preset **CB_Amps+Cabs** (fig. 7.14) uses two 3-Band Comps to split the audio into two Amp+Cab blocks. Try different amps and cabs to customize the sound. There’s usually enough processing power to add some effects. The effects that are available depend on the DSP power required by the Amp+Cab blocks.

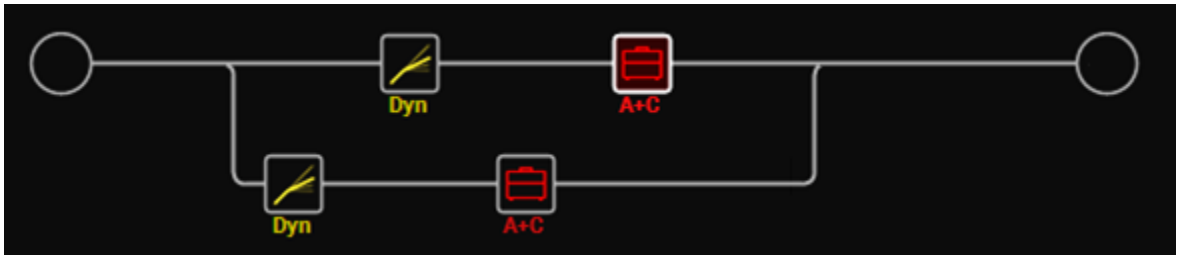


Figure 7.14 This preset replaces a single Amp+Cab with a Combi-Band setup that incorporates parallel Amps+Cabs.

To maximize the number of additional effects with amp replacement presets, the following techniques show how to stretch the available processing power. These techniques apply to Combi-Band setups with 2-Path devices, but they can also free up DSP power in 4-path Helix devices.

Share a Cab

Instead of using two individual Amp+Cab blocks, two amps can share a cab. The sound isn’t much different compared to using a cab for each. The tradeoffs are two amps can’t have different cabs, and any stereo imaging in the Mixer becomes mono after the cab. The preset **CB_A30FawnLead** (fig. 7.15) shares a cab to free up processing power for Dynamic, Distortion, and Reverb effects.

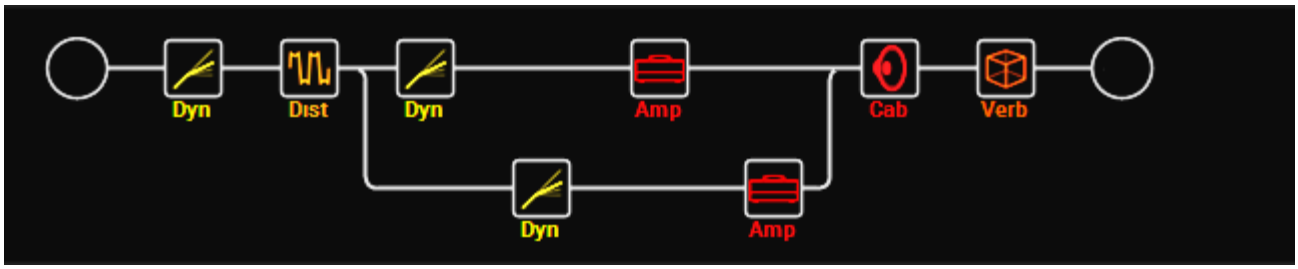


Figure 7.15 The two amps share a common cab to save processing power.

Path A’s A30 Fawn Brt handles the midrange. Path B’s A30 Fawn Nrm covers the low+high frequencies. The cab is the A30 Fawn’s usual 2x12 Blue Bell. The input has a Deluxe Comp followed by the Heir Apparent distortion. The stereo Double Tank at the end adds reverb and restores stereo imaging.

I mostly use this preset when playing “chunka-chunka” rhythm chords, combined with occasional forays higher up on the neck for single-note leads.

Share an EQ Cab

The Grammatico Amps require considerable DSP power. The preset **CB_GramatcoRhyth** (fig. 7.16) uses 3-Band Comps for Combi-Band splitting. For full stereo imaging, the amps share a stereo EQ-based cab and feed a stereo Simple Delay. (A version 3.50 Single cab could replace the EQ, but wouldn't preserve the mixer's stereo panning of the two amps.) A single EQ stage won't sound like a 1x12 Grammatico Cab, but it has its own sound, and it's easy to tweak the EQ to change the sound. Although stereo sounds best, the preset collapses well to mono.

The preset defaults to Wah off, but this preset illustrates another technique. Placing a Wah in one path gives a different wah sound, because now it's a parallel effect. It acts more like a tonal control. Furthermore, it gives four different characters, depending on which path it's in, and whether it's pre-amp or post-amp.

Figure 7.16 The Parametric EQ is emulating a cab.

Use an Amp+Cab in Each Path

The two paths in the preset **CB_2 Cabs** (fig. 7.17) use Amp+Cabs that don't need too much DSP power (Essex A15 and Brit P75 Brt). This frees up enough DSP for true stereo, and quite a few effects. Deluxe Comp and any of 16 distortions go before the amps. Post-amp effects are a "Tone Stack" stereo EQ and Legacy reverb.

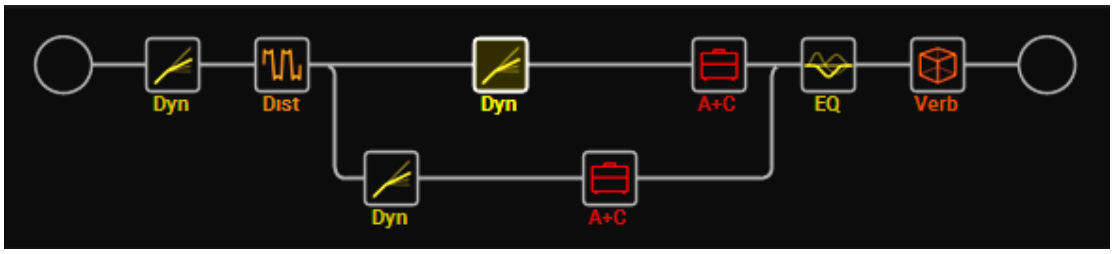


Figure 7.17 The efficiency of version 3.50 cabs allows using the 3-Band Comps for splitting. There's enough DSP power to add pre- and post-amp effects. The Mixer and post-amp effects maintain true stereo.

The preset has potential for customization:

- Delete the Deluxe Comp. Now you can choose from 20 Distortions.
- Delete the Reverb. All Distortions are available except Tone Sovereign, Vital Dist, and Legendary Drive. You can still add a stereo Simple Delay at the output.
- Delete the Deluxe Comp, Distortion, and Reverb. Now, any stereo Delay (except Cosmos Echo, Heliosphere, or ADT) is available as an output effect.

Use High-Efficiency Splits and Amps

Fig. 7.18 shows the preset **CB_Poly Capo**. It provides Combi-Band low tunings with full stereo. The Low and High Cut filter before the Poly Block conditions the signal for lower tunings. For higher tunings, raise the Low Cut frequency consistent with the best sound quality.

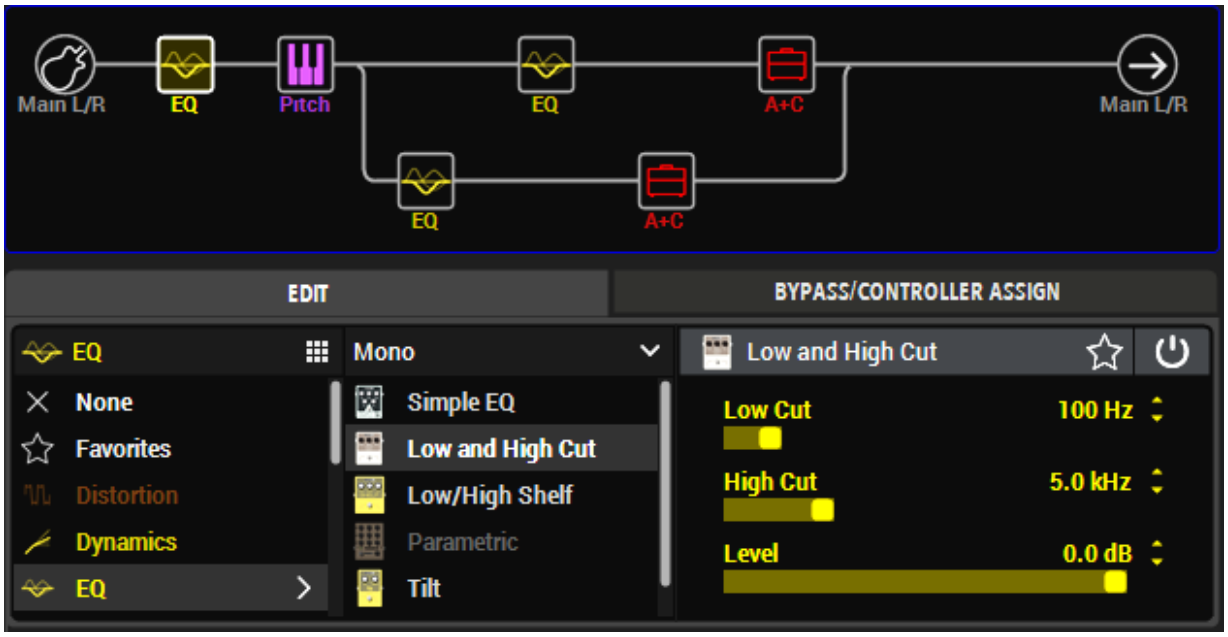


Figure 7.18 High-efficiency splitters and amps allow using the Poly Capo with a Combi-Band configuration.

With the same high-efficiency splitters and Amp+Cabs in fig. 7.18, you can use:

- Any Dynamics *and* Distortion effect.
- Any Legacy Reverb, or any mono or stereo reverb (except Dynamic Plate, Shimmer, and Hot Springs).
- Any stereo delay (except Cosmos Echo, Heliosphere, or ADT) in place of the Reverb.
- In place of the Reverb, you can insert any stereo Modulation effect other than Ampeg Liquifier, and any mono effect except for Poly Sustain.

Lead Echo

Before version 3.50, several presets in this eBook had to employ DSP-saving techniques. Those techniques are no longer needed. But rather than remove the presets, I updated them with version 3.50 cabs.

CB_CombiLeadEcho (fig. 7.19) uses a Combi-Band configuration for an echo-heavy lead sound. The input has optional Distortion. The output has a stereo Dual Delay. Previously, this preset required EQ-based cabs.

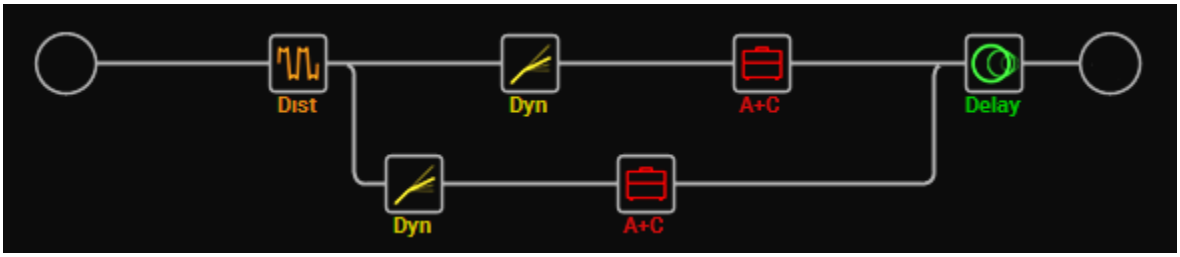


Figure 7.19 There's enough DSP power for a Scream 808 distortion at the input, and a Dual Delay (with the Chorus Modulation Mode enabled) at the output.

Tone Sovereign Crunch

Tone Sovereign has heavy DSP requirements. Legacy cabs allowed only 9 amp choices for a Combi-Band preset. The 3.50 cabs increased that to 35 amp options, so **CB_Soverhythm** (fig. 7.20) fulfilled my need for a US Princess-based, Combi-Band, crunchy stereo rhythm guitar sound.

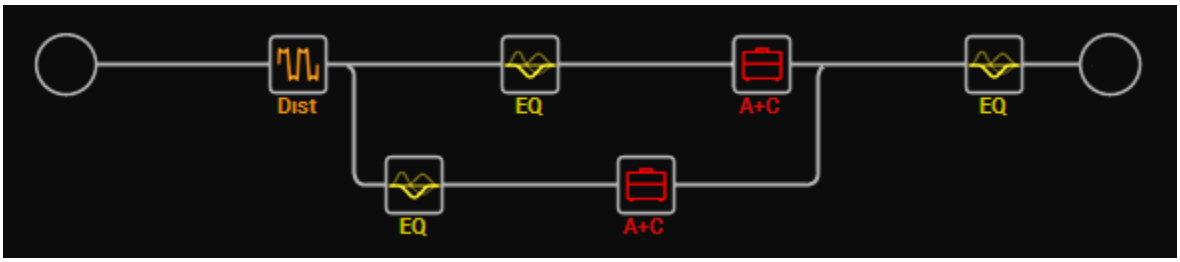


Figure 7.20 This preset modifies the Combi-Band concept to create a different tonal balance.

This preset shows another side of parallel path construction. Shelf EQ provides the Mid band, while Simple EQ provides the low+high bands. Not using the same type of EQ for splitting changes the tonal balance. Using different cabs for the two amps changes this further. Although both amps are a US Princess, one uses a 1x10 Princess Cab, while the other has a 2x12 Mail C12Q Cab. This gives the "humbucker-meets-rhythm-guitar" sound I wanted.

Wide Stereo Chords

Splitting with the Simple EQ requires the least amount of DSP power for Combi-Band amp replacement. The preset in fig. 7.21 harnesses the Placater Clean and Line 6 Ventoux for a somewhat understated crunch sound. Post-amp effects are stereo Hot Springs reverb and stereo Simple Delay.

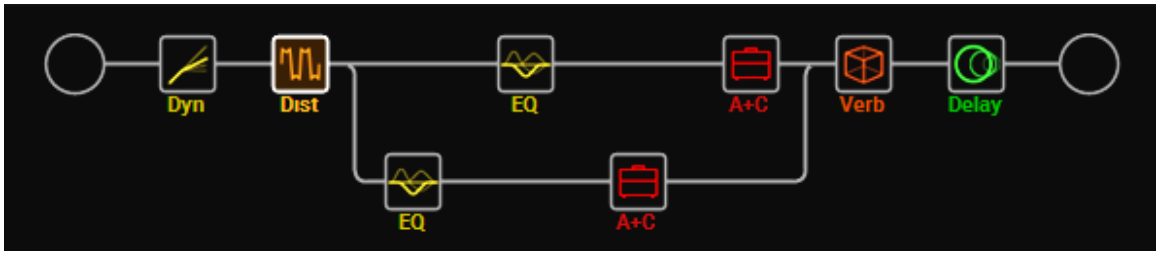


Figure 7.21 My favorite distortions with **CB_Stereo Chords** are Teemah!, Scream 808, and Stupor OD distortion.

This is a good preset for big, smooth power chords. The spacious stereo is ideal for recording, but the preset collapses well to mono for live performance.

Dirtier Splitting

Splitting with Shelf EQs creates more overlap between the bands. This gives a dirtier distortion sound than other splitting methods. **CB_DualPlexiJmp** (fig. 7.22) takes advantage of the more subtle splitting to retain some of the single-band Brit Plexi Jump sound. The 10 Band Graphic at the input conditions the sound before going into the amps.

There's enough DSP power left over to add a limited selection of effects, like a Deluxe Comp before the Split along with a stereo Simple EQ after the Mixer.

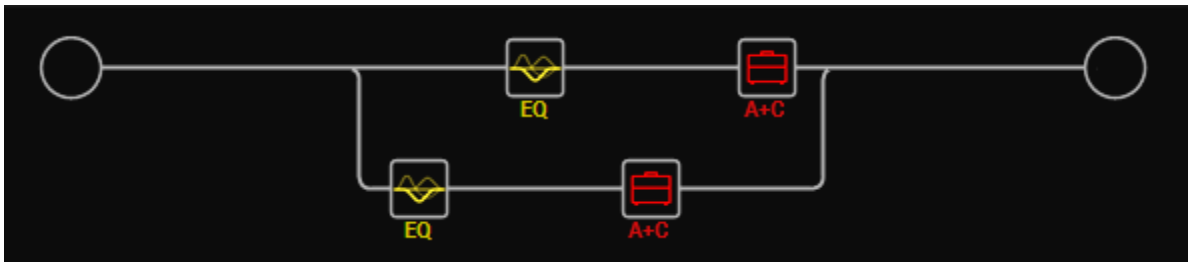


Figure 7.22 The Brit Plexi Jump is one of Helix's most DSP-hungry amps. **CB_DualPlexiJmp** combines two of them to create a big stereo sound that also collapses well to mono.

Stereo LCR Lead

Preset **CB_LCR_Lead** (fig. 7.23) splits the audio using Low/High Shelf filters. These require more processing power than Simple EQs, and the two frequency bands have more overlap. Using a Voltage Queen and A30 Fawn Brt Amp frees up enough DSP power to add any Distortion block.

This preset is designed for stereo, LCR-based mixing, so the two paths are panned hard right and left. The Simple Delay delays only the mid band. This increases the stereo separation.

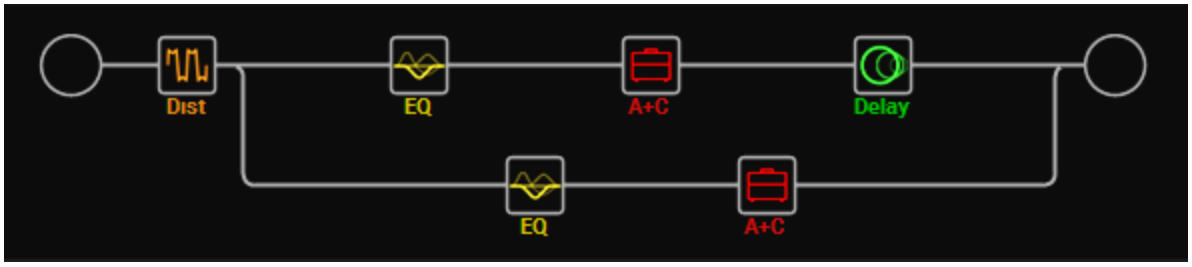


Figure 7.23 Amps that draw less power allow more DSP for effects. The Delay is still useful with mono stage setups.

Splitting with "Character" for More Coloration

CB_CleanPrincess (fig. 7.24) uses 10 Band Graphic EQs to split the bands. The splits are chosen to create a particular character, rather than provide an accurate frequency split. These EQs draw slightly less power than the Low/High Shelf EQs.

This preset also shows how to clean up an amp's sound by pulling way back on the Drive parameter (it's set to 1.0 for both amps). The sound has a bit of crunch and works well for rhythm guitar. To increase the intensity, enable the pre-split Kinky Drive distortion. The Deluxe Comp at the beginning evens out the dynamic response. With more extreme settings, it increases sustain.

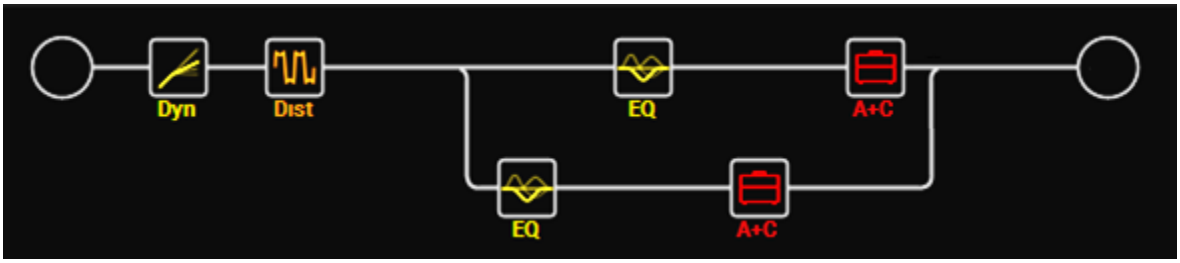


Figure 7.24 **CB_CleanPrincess** combines sweetness with a crunchy sound.

Preamps with Splits Save DSP Power

Combi-Band processing can also leverage Preamps to save processing power, as described in Chapter 3. Fig. 7.25 uses two Mandarin 80 preamps in the preset **CB_MandrinPreamp**. With preamps instead of amps, you can choose *any* mono Distortion at the input, and any stereo Delay, Modulation, Filter, Pitch/Synth, or Reverb block at the output. Using amps instead of preamps limits your choice of effects.

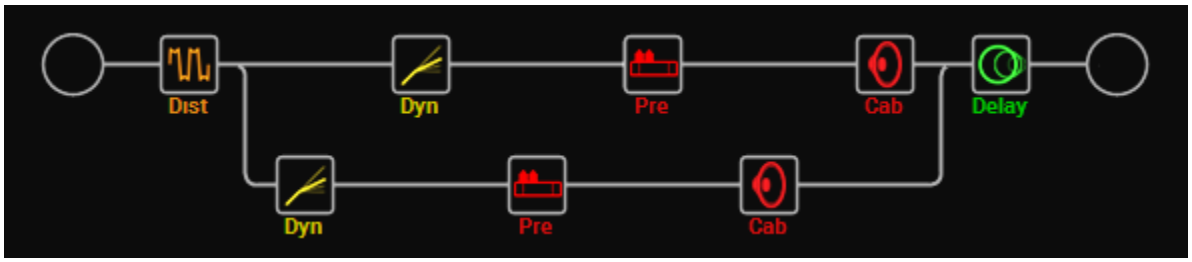


Figure 7.25 Using Mandarin 80 preamps instead of amps reserves much more DSP power for effects.

I like this preset a lot. It takes full advantage of the Mixer's stereo panning, is ideal for LCR mixing as well as conventional stereo, and collapses well to mono.

Use an EQ-Based Cab in Each Path

Before the version 3.50 cabs, presets like this saved DSP power so you could add effects. Although EQ-based cabs are no longer needed to save DSP power, they have their own sound.

The two paths in the preset **CB_EQ Cabs** (fig. 7.26) use EQ-based cabs that are optimized for each cab's associated frequency band. The Mixer can pan the two paths and alter their balance. The final stereo Simple EQ shapes the overall sound. A similar preset, **CB_EQ Cabs2**, shows how changing EQ settings alters the overall character. Even slight changes can make a big difference in the final sound.

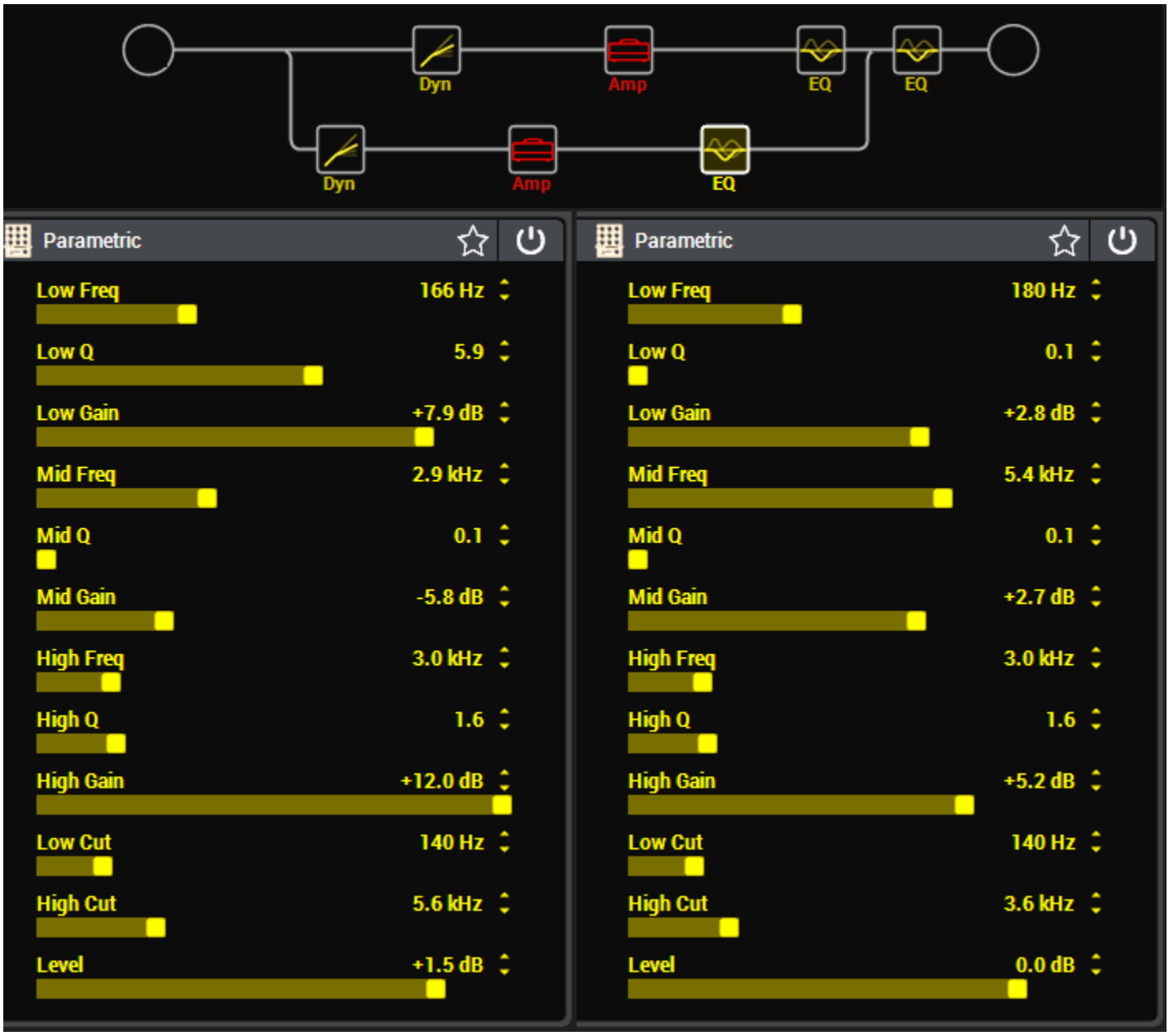


Figure 7.26 Replacing the Cabs with EQ provides a different sonic character. In the above image, the left parametric EQ is the cab for Path A (midrange). The EQ on the right is for Path B (low+high frequencies).

Customizing Amp Replacement Presets

These Combi-Band amp replacement presets are excellent candidates for customization. Although they use more blocks than single-amp presets, being able to tweak the sound deeply is often worth the tradeoff. Here are additional tips:

- Splitting techniques other than the 3-Band Comp require less DSP processing power.
- Although the 3-Band Comp is a great crossover for amp replacement presets, you can also take advantage of its compression options. For example, compress only the mid path by lowering the Mid Thresh and raising the Mid Gain. This gives a more present, sustained sound. Or, compress the Low band (lower the Lo Thresh, raise the Lo Gain) for a beefier sound.
- Experiment with altering the crossover frequencies for the various splitter options. You may want some bands to cover a wider range, or overlap.
- Use different amps and cabs for stacked sounds.
- A big advantage with Combi-Band presets is being able to edit amp parameters for specific bands. For example, alter Bias and Sag to alter the sounds of different bands.
- With amps that don't use much DSP processing, there's enough power to insert additional effects. This may require tradeoffs, especially with stereo vs. mono effects.
- In the studio, presets that preserve the Mixer's stereo image work well for LCR (left-center-right) mixing techniques. In addition to panning the paths hard left and hard right, try panning one amp to center, and the other hard left or right. This "tilts" the guitar sound to one side or the other.

Tech Talk: LCR (Left-Center-Right) Mixing

LCR mixing is a panning technique with all panpots set to either left, center, or right—nothing in between. It's appropriate for some musical contexts, but not others. Proponents of this mixing style claim several advantages:

- The resulting mixes sound very wide, because there's so much energy in the sides.
- It simplifies mixing decisions, because there are only three choices for stereo placement.
- Mixes translate well for those not sitting in stereo's "sweet spot," because the most important material is panned to the center.
- It forces you to pay attention to EQ and the arrangement, to ensure there's good differentiation among instruments panned hard left or hard right.
- If an LCR mix leaves "holes" in the stereo field, then you can use reverb or other stereo ambience to fill that space. As one example, stereo overhead mics on drums can pan hard left and hard right, yet still fill in a lot of the space in the center. Or, place reverb in a channel opposite of where a signal is panned.

Some engineers prefer LCR mixes for the reasons given above. However, LCR is not a panacea, and it also may not fit an artist's musical goal. Furthermore, some engineers consider a mix to be essentially a variation on mono, because the most important elements are panned to center. They don't want distractions that are panned hard left and right. Instead, they use what's on the left and right to provide a "frame" around the center.

Another consideration is that many people listen with headphones rather than component system speakers. Initially, LCR mixing may sound great on headphones due to the novelty, but it can eventually become disorienting. Then again, many people now listen to music on computer speakers. Any panning nuances are lost under those conditions, whereas LCR mixing can sound direct and unambiguous.

Chapter 8 | BUILDING LEAD PRESETS

The huge roster of effects, amps, and cabs allows for creating single-note lead sounds that span the range from traditional to novel. This chapter is about building your own lead presets.

One reason this eBook concentrates on techniques to create your own presets is because you can find a huge number of ready-to-go, user-rated presets at https://line6.com/customtone. Unlike the multiband presets covered in other chapters (and which are less common online), lead presets have the advantage of requiring only one or two amps and cabs. That leaves plenty of DSP power for effects. You then need to decide how to route them, using an optimum effects order.

Effect Routing Options

Helix and HX Stomp can insert effects in the three most common effects routings.

Series Routing

With this traditional pedalboard routing, one effect's output goes to another effect's input, whose output feeds another effect's input, and so on (fig. 8.1).

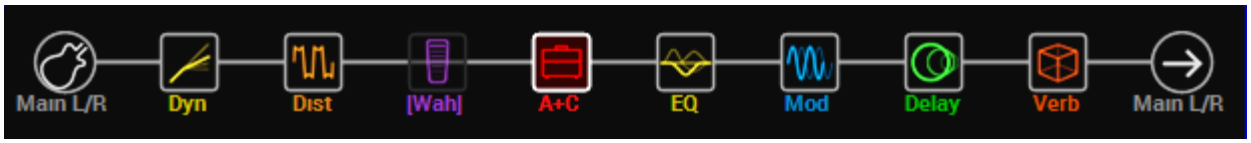


Figure 8.1 A typical series effects connection.

Parallel Routing

This splits the signal into two or more parallel paths (fig. 8.2). After processing the paths, the output Mixer re-combines them. See the Input Split section for more about splitting options—Y, A/B, Dynamic, or Crossover.

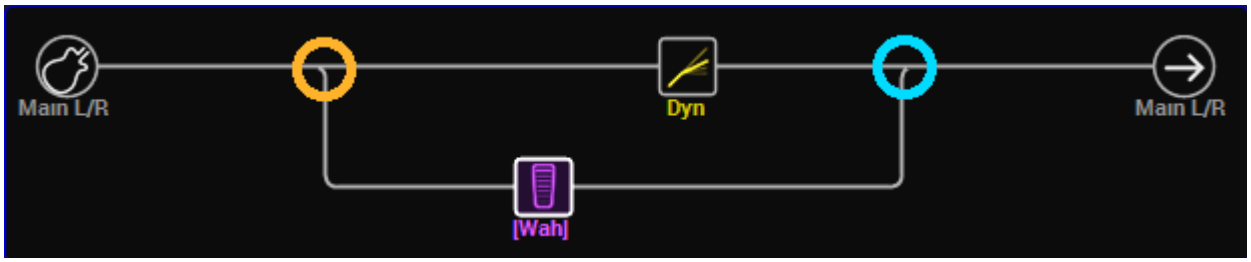


Figure 8.2 With bass, having the Wah in Path B doesn't thin out the main bass sound going through Path A. The orange circle represents the input Split. The blue circle represents the output Mixer.

Parallel-Series Routing

This combines the two approaches (fig. 8.3). The guitar or bass takes two separate paths, each of which has effects in series. Mixing them together gives the final sound.

Figure 8.3 Path A creates a distorted lead sound. Path B uses a pitch module to create a clean, octave-below sound.

The Mixer

This important block's parameters select the balance between parallel or parallel-series paths. For stereo applications, it also pans the paths. Fig. 8.4 shows the available Mixer controls.

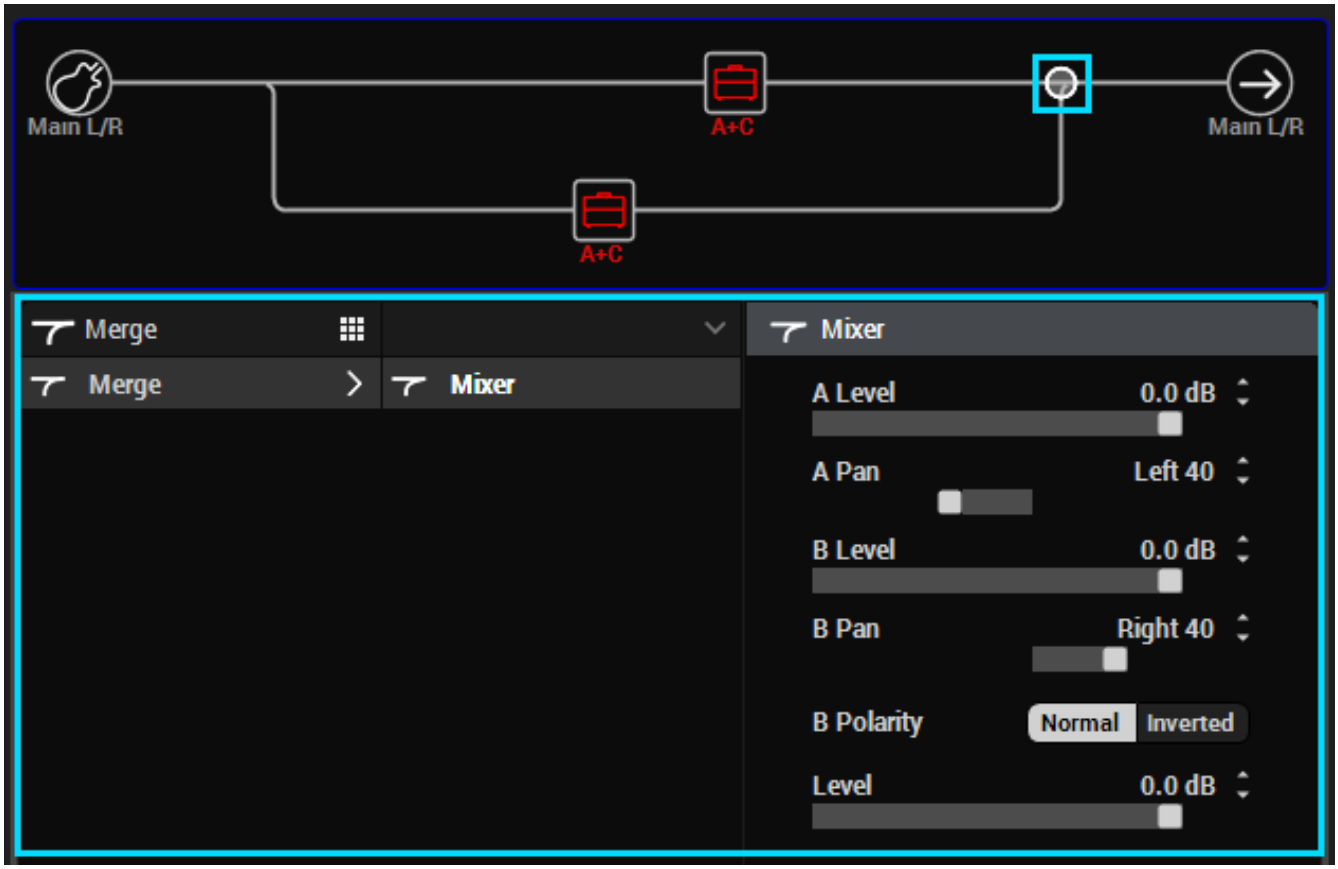


Figure 8.4 The Input Split is in Crossover mode. This creates high- and low-frequency paths that go through their own Amp+Cab blocks. The Mixer (blue outline) pans the two paths somewhat oppositely, mixes them together (each path has its own level parameter), then sends the mixed signal to the output.

- Path A (top) and Path B (bottom) both have **Level** and **Pan** parameters.
- The **B Polarity** switch changes the polarity (also commonly known as phase) when you select Inverted. We'll cover its practical applications later.
- **Level** adjusts the composite output of the two mixed paths.

Optimum Effects Chain Order

Actually…there is no optimum order, because it depends on your personal taste. However, the order listed below (from first to last in a series chain of effects) is a common starting point.

Gate

This dynamics processor is often the first effect. It can reduce hum and other noises before they get into other effects. Trying to remove noise *after* your signal goes through effects that amplify noise is difficult, because the noise has now become part of the sound. So, removing the noise will also remove audio you *do* want to hear. Helix includes a sophisticated Hard Gate effect, but in many cases it's not necessary. The Input block includes a gate, with Threshold and Decay parameters (fig. 8.5).

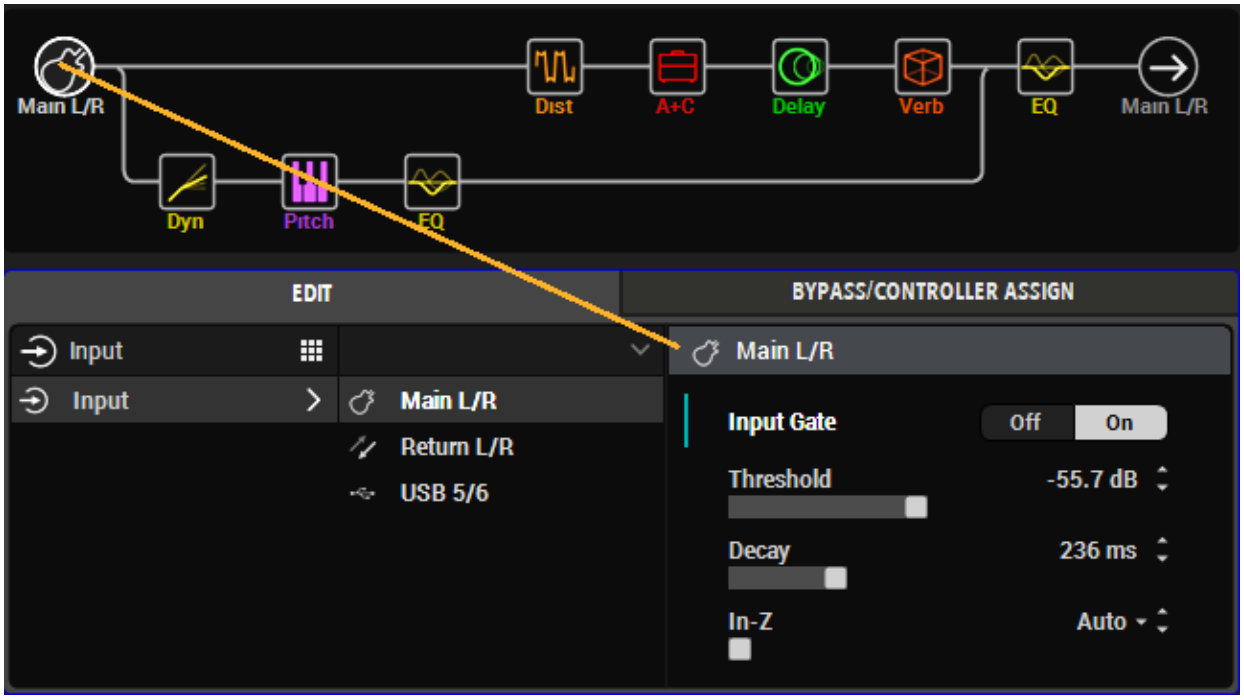


Figure 8.5 The Input block gate will handle most, if not all, of your gating needs.

Harmonizing/Synthesizing Effects

Pitch/Synth blocks *usually* work best with a clean guitar sound. Inserting effects like distortion or wah before them can interfere with proper performance. One exception is dynamics processors and EQ. These may be able to condition the sound going to the Pitch/Synth block, which could improve the effect's performance. Also, sometimes inserting these types of effects between an amp and cab works well. As always…experiment.

Envelope Filters

Blocks like the Mutant Filter, Mystery Filter, and Autofilter react to the guitar's dynamics. Distortion and Dynamics reduce the guitar's dynamic range, so envelope-dependent effects often precede effects that alter dynamic range.

Dynamics

Compressors control dynamics. Because they amplify low-level signals, compressors often insert early in the signal chain. Then, they don't amplify noise from prior effects. Dynamics processors can also create more consistent levels for subsequent effects (e.g., increase sustain with distortion).

Distortion

Distortion often benefits from processing the more consistent level produced by compression and limiting. Note that distortion also reduces dynamic range.

Amp+Cab

This determines the sound's overall tone and character.

Modulation Effects

Modulation effects, like Chorus, Phaser, Flanger, and Tremolo, have more impact when following distortion. Pre-distortion placement can mask a modulation effect's nuances (although this may be what you want).

Volume Pedal

When placed at the end of an effects chain, a volume pedal controls the overall output. However, you may want delay or reverb to continue after pulling back on the volume. This can't happen if the volume pedal is at the end of the effects chain. Many pedalboard setups insert the volume pedal before reverb or delay. They insert toward the end of a chain anyway.

Delay

Delaying a distorted signal sounds more focused than distorting a delayed signal. Delays normally follow the amp and cab.

Reverberation

This represents a space for your guitar and other effects, so it goes last. It's like how a room is the final "processor" for a physical amp's sound.

...But Effects Order "Rules" Were Made to Be Broken

The above effects chain represents a basic, default order. But there are as many exceptions and alternatives as there are guitar players seeking a unique sound.

EQ

Regarding EQ's placement within an effects chain, it's a general-purpose effect that can be useful in multiple places. For example, you might want EQ before an amp to reduce the highs for a less "harsh" sound, or at the end of the chain, as a final tone control.

EQ Before/After Compression

Compression can serve different purposes. Placing compression after EQ gives a more consistent level, and increases sensitivity to certain frequency ranges. For example, if the EQ creates a massive midrange boost, these frequencies will be compressed before other frequencies.

However, post-EQ compression "undoes" some of the effect of boosting or cutting EQ, because compression reduces peaks and brings up softer levels. For EQ to have the greatest impact, place compression before it.

Wah Before/After Distortion

Wah has the most pronounced effect when placed after distortion, because it thins the distorted sound. This is common in funk and R&B, which usually aren't as distortion-centered as rock. When placed before distortion, wah produces a more ripping, cutting kind of sound. This is more common with rock, because musicians initially placed wahs in front of amplifiers that often added distortion.

Envelope Filter Before/After Compression

A highly dynamic signal sweeps an envelope filter over its widest range. This produces the most pronounced effect. However, envelope filters generally thin out the sound. At the low end of the range, the sound may be muddy. At higher frequencies, the sound may be so thin the volume drops too much. Although envelope follower parameters can restrict the envelope follower's range, it can be difficult to pick your strings with the precision needed to obtain the desired results. An envelope follower preceded by a little compression may better accommodate your playing style.

Distortion Before/After EQ

EQ after distortion alters the distorted sound's timbre. But similarly to how using EQ before compression can give more frequency-sensitive dynamics, EQ before distortion causes boosted frequency ranges to distort more readily than other ranges. For example, boosting a guitar's upper midrange will cause single-note solos higher up on the neck to distort at lower levels than notes played lower on the guitar.

I often insert EQ both before *and* after distortion. Before distortion, rolling off the high frequencies creates a creamier sound, while reducing lows avoids low-frequency "bloat." After distortion, a slight upper-midrange boost can improve articulation. However, boosting too much increases harshness.

Flanger Before/After Distortion

Flanger before distortion dilutes the effect, but you may prefer how a high-resonance flanging sound "cuts" through the distortion.

Flanger Before/After Dynamics

Flangers can generate strong amplitude peaks and deep valleys. Following the flanger with compression restricts the dynamic range and tames the peaks. However, too much compression reduces the effect's intensity.

EQ Before/After Modulation

Inserting EQ before modulation can optimize the timbre to work well with the modulation effect. Inserting EQ afterward modifies the modulated sound.

Compression Before/After Delay

Compression after delay reduces the delay's effectiveness, because as the delays fade out, the compression tries to raise their levels. Compression before delay affects the signal feeding the delay, not the delay itself.

Delay Before/After Reverb

I usually prefer delay before reverb, because then the delays feel like they're in a room. For a more diffused sound, try reverb before delay. Or, insert them in parallel (see next) to differentiate each effect more.

Common Parallel Effects Applications

Before creating presets, consider some of these parallel effects applications:

- **Wah or envelope-controlled filter.** Filtering won't thin out the main channel's sound when placed in parallel with the channel it's processing.
- **Bass.** Keep the full, round bass sound in the main channel. Use processors like distortion, chorus, wah, etc. as parallel effects. This preserves the low end.
- **Add more variety.** Placing two modulation effects in parallel, and setting the modulation rates to different speeds, gives a more randomized sound than one effect with a single LFO.
- **Use parallel envelope filters.** Programming slightly different settings can sound more animated than a single envelope filter.
- **Mix heavy compression with a dry signal path.** Parallel compression combines the compressed signal's sustain with the dry signal's dynamics.
- **Use panning to your advantage.** With stereo, insert different cabs in parallel, then pan the two paths somewhat left and right of center. A rhythm guitar part panned through two amps is almost like having two guitars, but with the focus of a single guitar part.
- **Use effects with different, tempo-synched rhythmic values.** Placing tremolo or delay in parallel, and synching the effects to different rhythms, gives a feeling of motion. With stereo effects, this also spreads the image more.
- **Unusual time delay effects.** Insert two vibrato effects (with processed sound only), set for a slow speed, in parallel. Stereo produces novel panning-like effects in headphones.

Mono Lead Effects for Live Performance

Let's create some basic lead presets, designed for live performance. We'll start with a preset that provides a "test bed" for preset development. It's the *Free Files > Chapter 8 Preset* folder.

Create Your Own Sounds

Helix includes several effects categories: Distortion, Dynamics, EQ, Modulation, etc. Fig. 8.6 shows the preset **Lead Preset Dev**, a typical chain for lead sounds. This includes the most important Helix categories, except for Pitch/Synth and Filter. We'll cover variations later, but this chain makes it easy to evaluate different effects and decide how they fit into your sound.

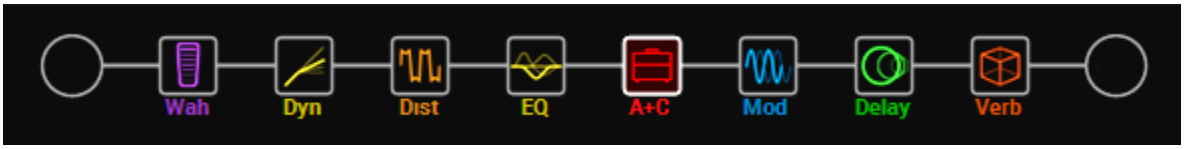


Figure 8.6 Test setup for creating lead presets.

This test setup makes it easy to substitute different distortions, dynamics, wahs, delays, etc. to hear how they affect the tone. The Voltage Queen Amp+Cab has average DSP power requirements. To audition an effect that's grayed-out because there isn't enough processing power to run it, substitute a Soup Pro for the Voltage Queen. This frees up DSP power to accommodate more effects. To free up more DSP, also remove the reverb.

The effects in **Lead Preset Dev** are gain-staged, so you can enable and bypass the effects and not hear significant changes in the perceived level. When comparing effects, adjust the gain so there isn't a big volume variation. Levels influence how people perceive sounds. If a sound is louder compared to another sound, the louder one is often considered as "better."

Time to start exploring…

Check Out the Amps

The Amp+Cab blocks are Helix's heart and soul. Bypass the other effects to hear the pure Amp+Cab sound, then call up different models to audition their characters. Vary the parameters to learn their capabilities (not all amps have the same parameters). There's detailed information on some of these parameters under Amp Blocks. Here's a summary:

- For more distortion, increase **Drive**. Reduce it for more articulation. The amp's defaults are based on typical settings for the physical amps being modeled. However, you'd probably change a physical amp's settings, so you may want to change a virtual amp's settings. I prefer less drive with dense arrangements. Excessive distortion can clutter the sound.
- **Channel Volume** sets the overall level. This parameter doesn't influence tone. It's a good choice for balancing levels when choosing different amps for similar presets.
- **Master** volume does influence tone. The higher the setting, the more other settings influence the overall sound. To set your tone, edit this in conjunction with other amp parameters. To set the level, adjust Channel Volume.
- Whoever at Line 6 decided to include **Sag** deserves applause. The results of changing it are unpredictable and depend on the amp. Minimum sag gives a tighter sound. Maximum may sustain the sound more, but too much may detract from the excitement that happens with minimum sag. For my multiband presets designed for chords, I usually leave sag at minimum. Leads can benefit from some types of sag to produce a more even, sustained tone, with more touch sensitivity.
- **Hum** and **Ripple** can add authenticity to amp sounds. I always tried to engineer hum out of circuits, so I have no desire to add any. However, ripple sometimes adds more animation to a sound. I turn it up on a case-by-case basis.
- **Bias** is important. At minimum, the distortion is less focused. At maximum, it sounds warmer. I often end up about 2/3 – 3/4 of the way up for leads, but adjust this by sweeping the slider from one end to the other while playing. This helps find the sweet spot. The same is true of **Bias X**.

Dealing with Option Overload

Helix offers so many options, it's easy to be tempted into thinking you're just a few tweaks away from getting an *even better* sound. But avoid becoming too distracted. Given Helix's sheer number of amps, miking options, cabinets, effects, and more, the possible combinations are at least in the hundreds of thousands, if not millions. This is especially true because as you try different cab and virtual miking options, the results may not be consistent—a perfect mic choice for one amp might not be the best choice for a different amp.

If you're prone to option overload, try the following steps to narrow down your choices. This procedure is just one possible strategy:

1. Pick an Amp+Cab that sounds close to what you want.
2. Try out different cabs and choose the one that sounds best.
3. Next, try different Mics, then choose your favorite.
4. Edit the cab's Distance and Position (and with Legacy Cabs, Early Reflections) for the best sound.
5. Now, run through the amp options to see if one sounds better.

Avoid the temptation to experiment further—if you have a good sound, start playing! Although musicians have specific opinions about sound and tone, the average listener does not. People want to hear music…so give the people what they want. They're interested in the notes you play, whether they enhance the song, and if the tone is pleasing.

Lead Presets with the v3.70 Amps

I wanted v1.4 of this book to get more into leads, and the v3.7o amps provided a great reason to come up with some lead-oriented presets. In the section Amp Blocks in Version 3.70, with Presets, several of the amps are lead-friendly (some of which also work for rhythm). Each preset has a description that de-constructs the preset, explains which blocks were chosen, and covers some of the parameter values used in the blocks.

You'll find the presets in *Free Files folder > Chapter 2 Presets folder > 3.70 Amp Presets.*

Chapter 9 | 3- AND 4-BAND PROCESSING

4-Path Helix effects can create entirely new sounds, with nearly infinite programming possibilities, by using 3- and 4-band multiband processing.

Multiband processing is ideal for processing chords with distortion. Splitting the guitar into frequency bands and processing them individually allows optimizing the amp and cab for each frequency band. This enhances definition and articulation. You can also add effects for some bands, but not others.

Overview: Multiband Processing Options

We covered bi-amping in Chapter 5, and Combi-Band processing (a workaround for doing 3-band processing in a 2-Path Helix device, like HX Stomp) in Chapter 7. This chapter is relevant only with 4-Path Helix devices, because you need as many sub-paths as bands you want to process. 2-Path devices have only two sub-paths.

Multiband processing is an *alternative* to traditional processing techniques, not a replacement. Nonetheless, people are often blown away when they hear the clarity of multiband processing. To experience immediately why this is a cool technique, skip ahead to Practical Examples and Free Presets, and load some sounds.

3-Band Processing

3-Band Processing splits the audio into **low**, **mid**, and **high** bands. One Path processes two bands. The other Path processes the third band (fig. 9.1). All paths connect to the same input, and usually feed the same output.

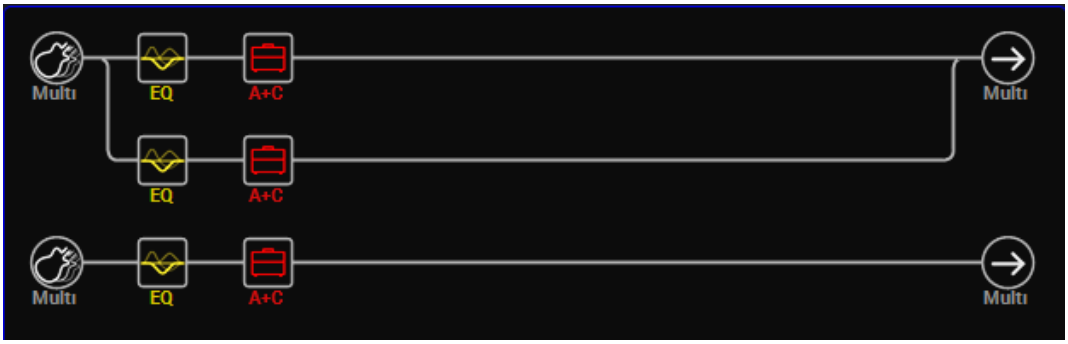


Figure 9.1 Typical 3-Band multiband processing setup.

Each path starts with a block that restricts the frequency range to a particular frequency band. The Input Split's crossover isn't useful, because it splits Helix's frequency range into two ranges. It can't split into *three* ranges.

In fig. 9.1, the top parallel sub-paths paths provide two of the ranges. For example:

- The EQ in the top path filters out the highs and mids, leaving only the low frequencies.
- The EQ in the middle path filters out the highs and lows, leaving only the mid frequencies.
- The EQ in the bottom path filters out the lows and mids, leaving only the high frequencies.

A Mixer combines Path 1's two parallel sub-paths. There's no rule about which path should handle a particular frequency band. You might want to use Path 1 to process the mids and highs, the lows and mids, or the lows and highs. What works best depends on the preset's goals. We'll cover some options later.

4-Band Processing

4-Band processing splits the guitar into **low**, **lower mid**, **upper mid**, and **high** bands. Again, input blocks isolate the specific frequency bands (fig. 9.2). Mixers combine Path 1 and 2's parallel sub-paths.

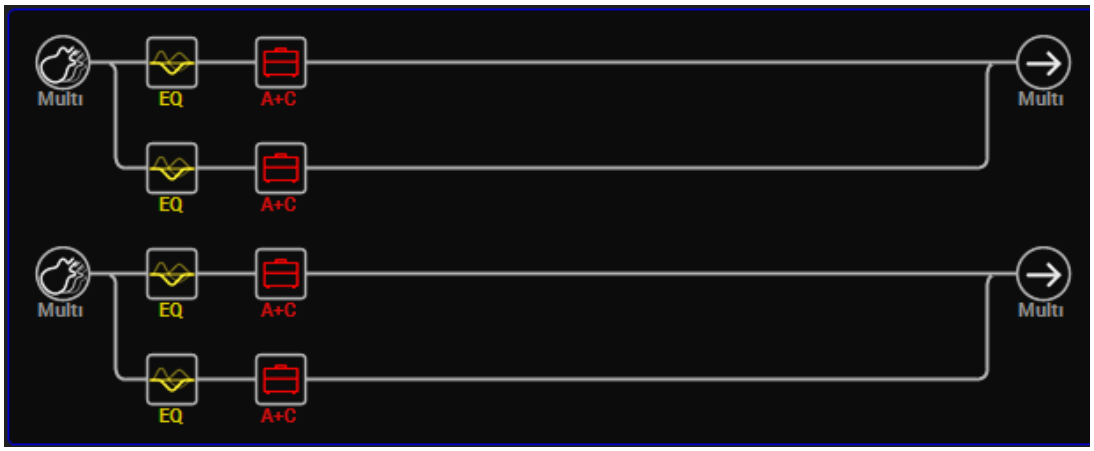


Figure 9.2 Typical 4-Band multiband processing setup.

Multiband Processing Tradeoffs

Although multiband processing can deliver exceptional sound quality, there are tradeoffs:

- Amps require significant DSP power, so there's less DSP power for other effects. 4-Band presets work best as "amp replacements" that give a more defined, articulated sound than single-band amps.
- Compared to single-band setups, multiband processing delivers a smoother, more focused sound with chords—i.e., playing multiple notes at different frequencies. With single-note leads, at any given moment a single note plays in only one band of frequencies. Lead presets don't always benefit as much from multiband processing.
- Unlike bi-amping, Helix doesn't have an Input Split that handles 3- or 4-Band processing. Each path needs a dedicated input block to narrow the frequency response to a specific range.
- More bands mean more parameters to edit (it took me a year to develop the Craig Anderton's Amazing Multiband Helix Presets pack). To get you started, the *Free Files* > *Chapter 9 Presets* folder includes ready-to-go multiband presets. You can then customize them for your own applications.
- Creating presets requires planning. For example, in a 3-Band preset, two bands are in parallel and one band is separate. The separate path would be best for the band that requires the most effects, because the path's DSP power isn't shared by two parallel paths.
- Creating or even optimizing multiband presets is *not* easy. The many variables and options require patience. Changing just *one* parameter in *one* amp can make a major sonic difference. But once you optimize a multiband preset, you'll have sounds you can't get any other way.

Multiband Processing Techniques

We'll cover several ways to split audio into 3 or 4 bands. The tradeoffs involve DSP power, tone, and precision (how accurately the bands split the frequencies). This is not a "good-better-best" situation. Sometimes, a less precise splitting method gives the desired sound.

3-Band Template Preset with the Most Accurate Splitting

The most accurate way to split into three bands uses the 3-Band Comp. As mentioned, defeating compression in the 3-Band Comp transforms it into a three-band crossover. We covered a similar application previously for Combi-Band splitting (see Create Combi-Band Paths with the 3-Band Comp). However, the Combi-Band technique is a workaround, because the high and low bands share a single path. A 4-Path Helix doesn't have that limitation. Each frequency band can have its own path.

Fig. 9.3 shows how to configure the 3-Band Comp block for isolating a band of frequencies. The block has the same settings for all three bands, except:

- For the low band, Lo Gain is 0.0. Mid and Hi Gain are -60.0 dB.
- For the mid band, Mid Gain is 0.0. Lo and Hi Gain are -60.0 dB.
- For the high band, Hi Gain is 0.0. Lo and Mid Gain are -60.0 dB.

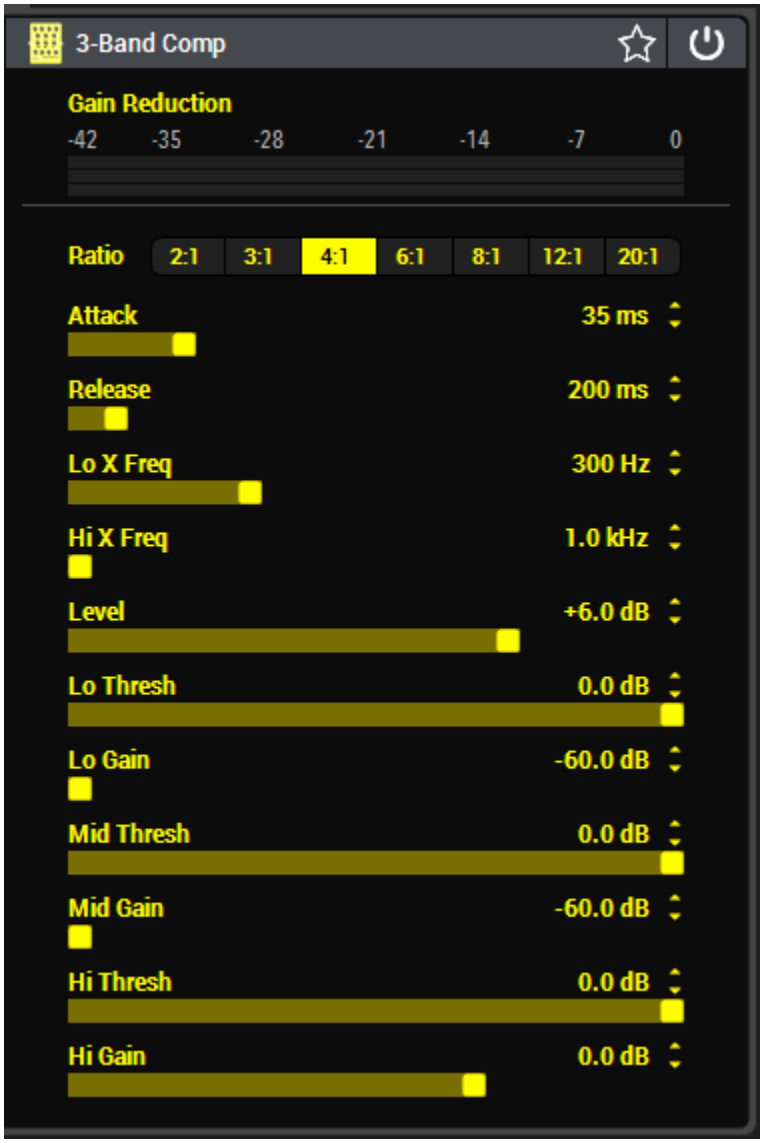


Figure 9.3 3-Band Comp settings to split audio into three bands.

The *Chapter 11 Favorites* folder includes blocks that filter the low, mid, and high bands (**3Bd_LowSplitCom.fav**, **3Bd_MidSplitCom.fav**, and **3Bd_HiSplitComp.fav** respectively). Insert these at the beginning of paths to create frequency bands. Customizable parameters are:

- **Low X Freq** and **Hi X Freq**. These edit the frequency ranges covered by the three bands.
- **Level**. Because the block at the path's beginning filters out audio, there's less level than usual going into the post-EQ effects. The Level parameter compensates for this. Furthermore, if an amp follows the block, you can increase a 3-Band Comp's Level to push more signal into an amp. Or, decrease Level to clean up the sound.

The *Free Files* > *Chapter 9 Presets* > *Templates* folder has three, 3-Band Comp-based templates (fig. 9.4). The Vol (Gain) blocks in the templates simplify tweaking. Set them to 0 gain. Enabling the Gain block mutes the associated path. To edit a path without being distracted by other paths, mute the other paths. After creating the preset, delete the Vol (Gain) blocks. For more information, see Create a Mute Button.

In all 3-Band templates, Mixers pan A to Left 50 and B to Right 50. The Dual Cabs pan full left and right.

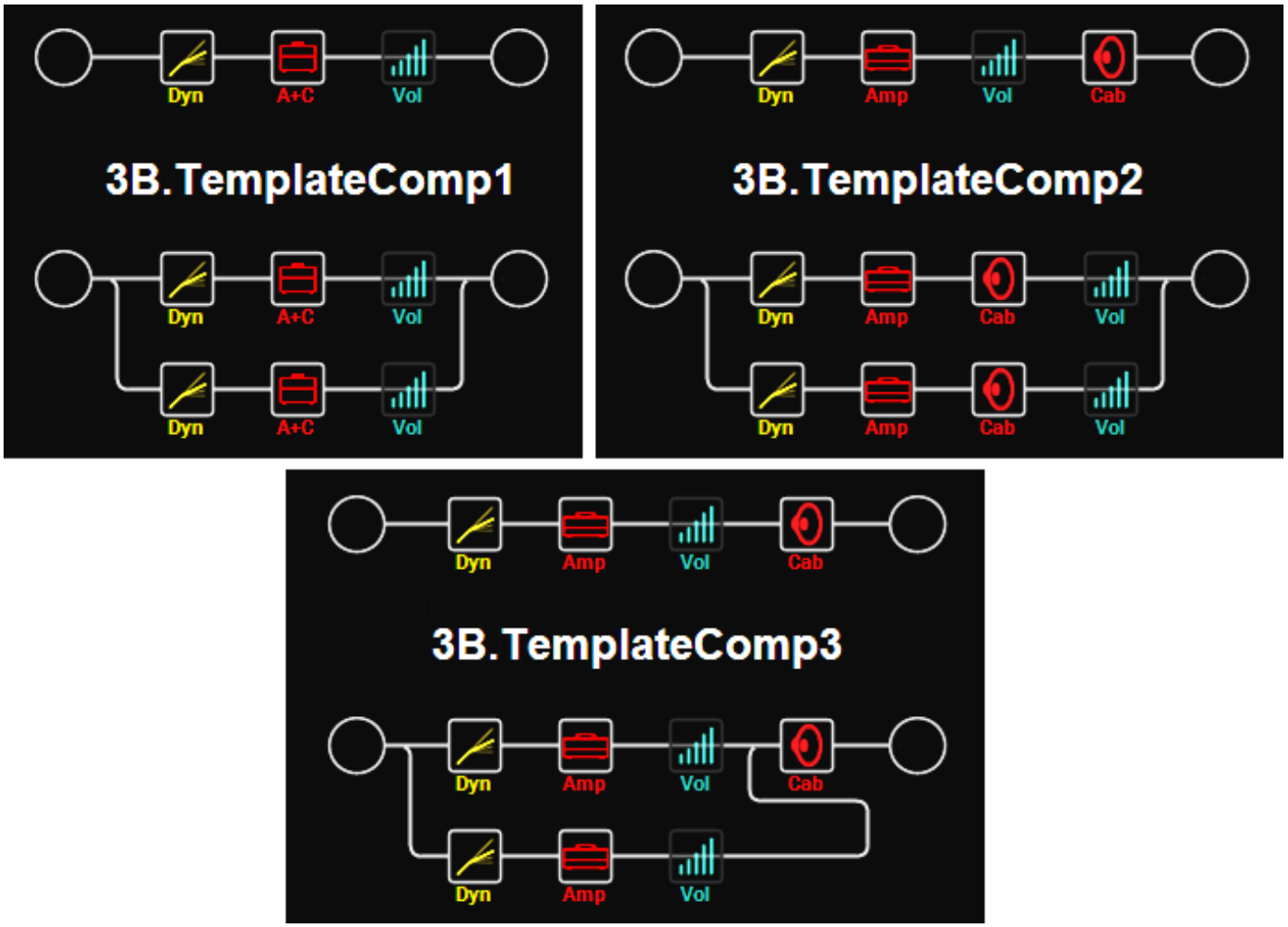


Figure 9.4 3-Band templates based on the 3-Band Comp.

- **3B.TemplateComp1** inserts an Amp+Cab in each path. With version 3.50 cabs, Path 2 can load parallel instances of all Amps+Cabs except for PV Vitriol Crunch, PV Vitriol Lead, and Brit Plexi Jump.
- **3B.TemplateComp2** dedicates a separate amp and separate cab to each path. This simplifies inserting Cabs that have been saved as Favorites, and allows using a Dual Cab in Path 1. Or, combine the Path 2 Amp and Cab blocks into Amp+Cab blocks to free up two blocks. Amp and cab processing power considerations for Path 2 are the same as for the template **3B.TemplateComp1**. Using a Dual Cab in Path 1 requires additional DSP power compared to using a Single cab.
- **3B.TemplateComp3** saves processing power by sharing a Dual Cab between Path 2's parallel paths. This maintains any stereo panning in the Mixer. Replacing the Dual Cab with a Single Cab saves DSP but negates Mixer panning. The Path 1 Dual Cab creates a stereo output. Alternately, combine Path 1's Amp and Cab into an Amp+Cab block to use one less block and save some DSP power. Note: You can't insert two PV Vitriol Crunch or PV Vitriol Lead Amp+Cabs in parallel. Also, inserting either one limits your choices for the other amp that's in parallel.

3-Band Template Preset that Uses Less Processing Power

Splitting with 10 Band Graphic EQs requires less than half the processing power of using 3-Band Comps. The same comments about deciding which splits to use in parallel sub-paths, and using Gain blocks as mute buttons for paths, apply here.

The **3B.TemplateGEQ1, 3B.TemplateGEQ2**, and **3B.TemplateGEQ3** templates have the same configuration as fig. 9.4. With these 10 Band Graphic EQ-based templates, the Low, Mid, and High 10 Band Graphic blocks replace the Lo, Mid, and Hi 3-Band Comp blocks. Similarly to the 3-Band Comp-based templates, the Level parameters for the 10 Band Graphic EQs can affect amps that follow these blocks. Pushing more or less signal into the amp is equivalent to changing the amount of overdrive.

Fig. 9.5 shows the 10 Band Graphic settings for the three bands. The Favorites folder includes the blocks **3Band_LowSplitGEQ.fav**, **3Band_MidSpltGEQ.fav**, and **3Bd_HiSplitGEQ.fav** for the low, mid, and high bands respectively.

Figure 9.5 Using Graphic EQ blocks to do frequency splitting requires less processing power than 3-Band Comps. Although the splitting provides less isolation among the bands, you may prefer this for some sounds.

3-Band Template Preset that Uses the Least Processing Power

Splitting with Simple EQs requires the least processing power. This can make the difference between squeezing in a few extra effects or not. The tradeoff is more overlap among the frequency bands.

The templates **3B.TemplateSimp1, 3B.TemplateSimp2**, and **3B.TemplateSimp3** have the same configuration as fig. 9.4. With these Simple EQ-based templates, the Low, Mid, and High Simple EQ blocks replace the Lo, Mid, and Hi 3-Band Comp blocks.

Fig. 9.6 shows the Simple EQ settings for the three bands. The Favorites folder includes the blocks **3Bnd_LowSpltSimp.fav**, **3Bnd_MidSpltSimp.fav**, and **3Bnd_HiSpltSimp.fav** for the low, mid, and high bands respectively.

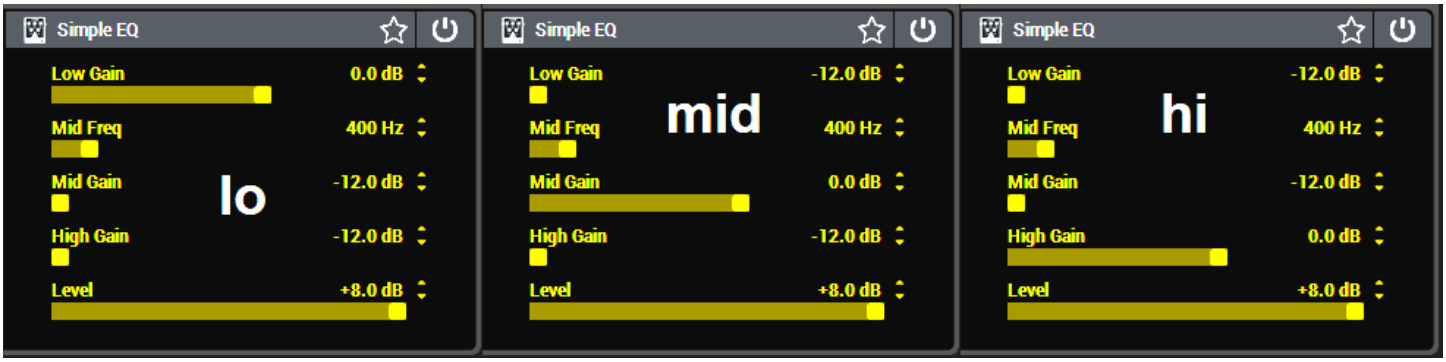


Figure 9.6 Simple EQ settings for low, mid, and high bands.

All three types of splits are useful. For example, the **3B.SimpEQSplit** preset (fig. 9.7) in *Free Files > Chapter 9 Presets* splits the audio using Simple EQs. For Path 1, there's enough processing power to add several effects with a Moo)))n Amp (including DSP-hungry ones like the Tone Sovereign), compressor, a Dual Cab, and stereo modulation/reverb/delay. The other splitting methods can't add all these stereo effects with this Amp.

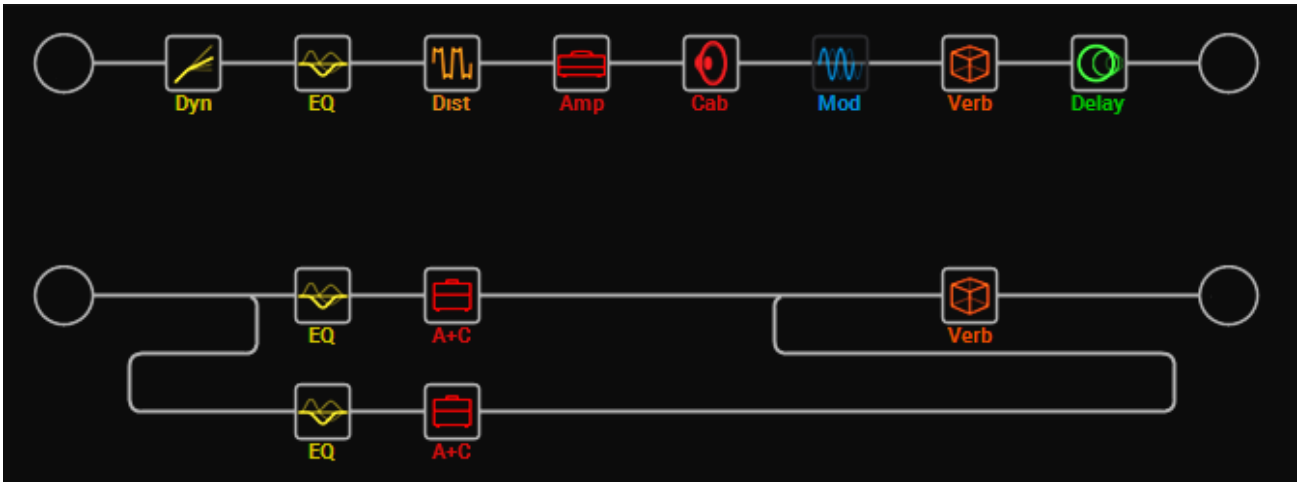


Figure 9.7 **3B.SimpEQSplit** preset. There's enough DSP power to include multiple effects.

4-Band Template Preset with the Most Accurate Splitting

3-Band Comps can create four splits. However, the Hi X Freq can't go lower than 1 kHz, which is not ideal. Fig. 9.8 shows possible four-band settings. The low and low mid frequency splits use the block on the left, high mid and high frequency splits use the block on the right. These settings are typical:

- Low Band: Up to 300 Hz (Lo X Freq). Lo Gain = 0.0. Mid and Hi Gain = -60.0 dB.
- Low Mid Band: 300 Hz to 1 kHz (Hi X Freq). Mid Gain = 0.0. Lo and Hi Gain = -60.0 dB.
- High Mid Band: 700 Hz (Lo X Freq) to 3.0 kHz (Hi X Freq). Mid Gain = 0.0. Lo and Hi Gain = -60.0 dB. (Having the High Mid band's Lo X Freq dip down to 700 Hz helps make up somewhat for the Low Mid's Hi X Freq not being able to go below 1 kHz.)
- High Band: Above 3.0 kHz (Hi X Freq). Lo and Mid Gain = -60.0. Hi Gain = 0.0 dB.

The Favorites folder includes **4Bd_LoSplitComp.fav**, **4Bd_LoMidSptCom.fav**, **4Bd_HiMidSptCom.fav**, and **4Bd_HiSplitComp.fav**. These cover the low, low mid, high mid, and high bands respectively.

Fig 9.9 shows 4-Band templates based on the 3-Band Comps. Available DSP power constrains what you can insert in the templates other than blocks for splits, amps, and cabs.

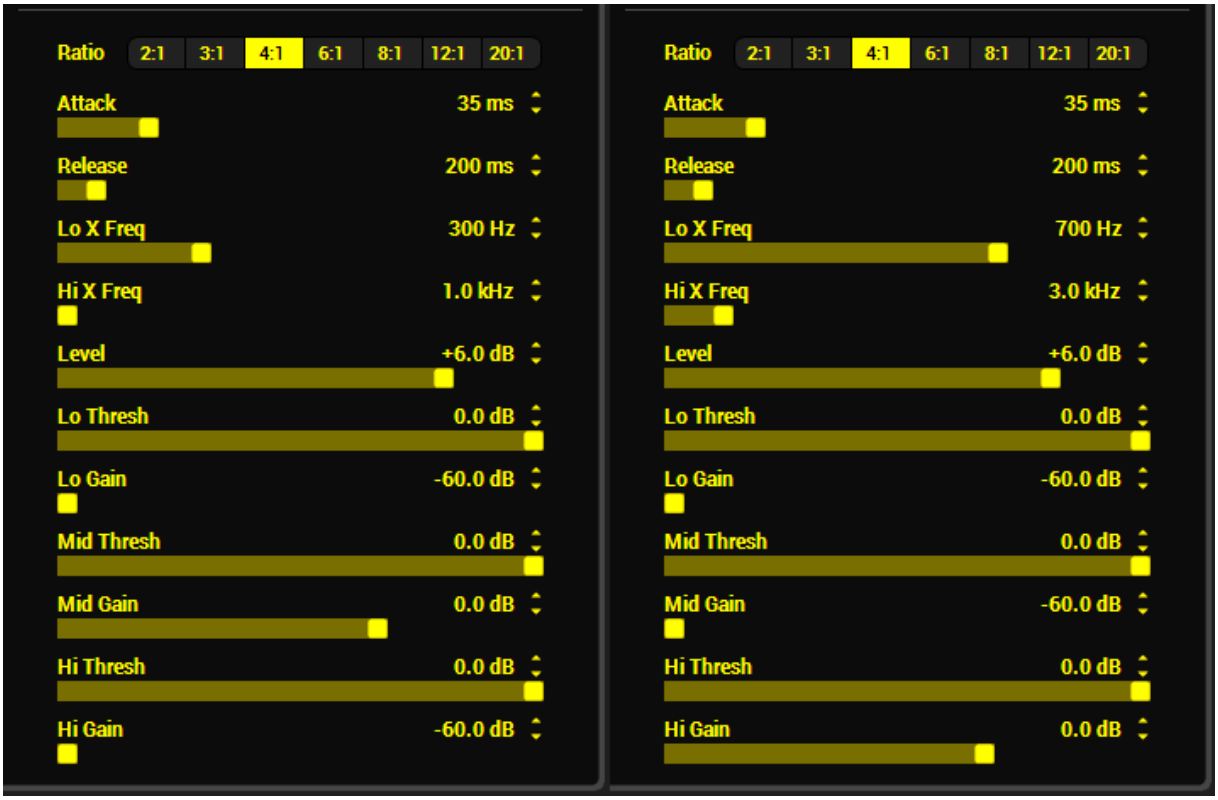


Figure 9.8 3-Band Comp settings for 4-Band templates.

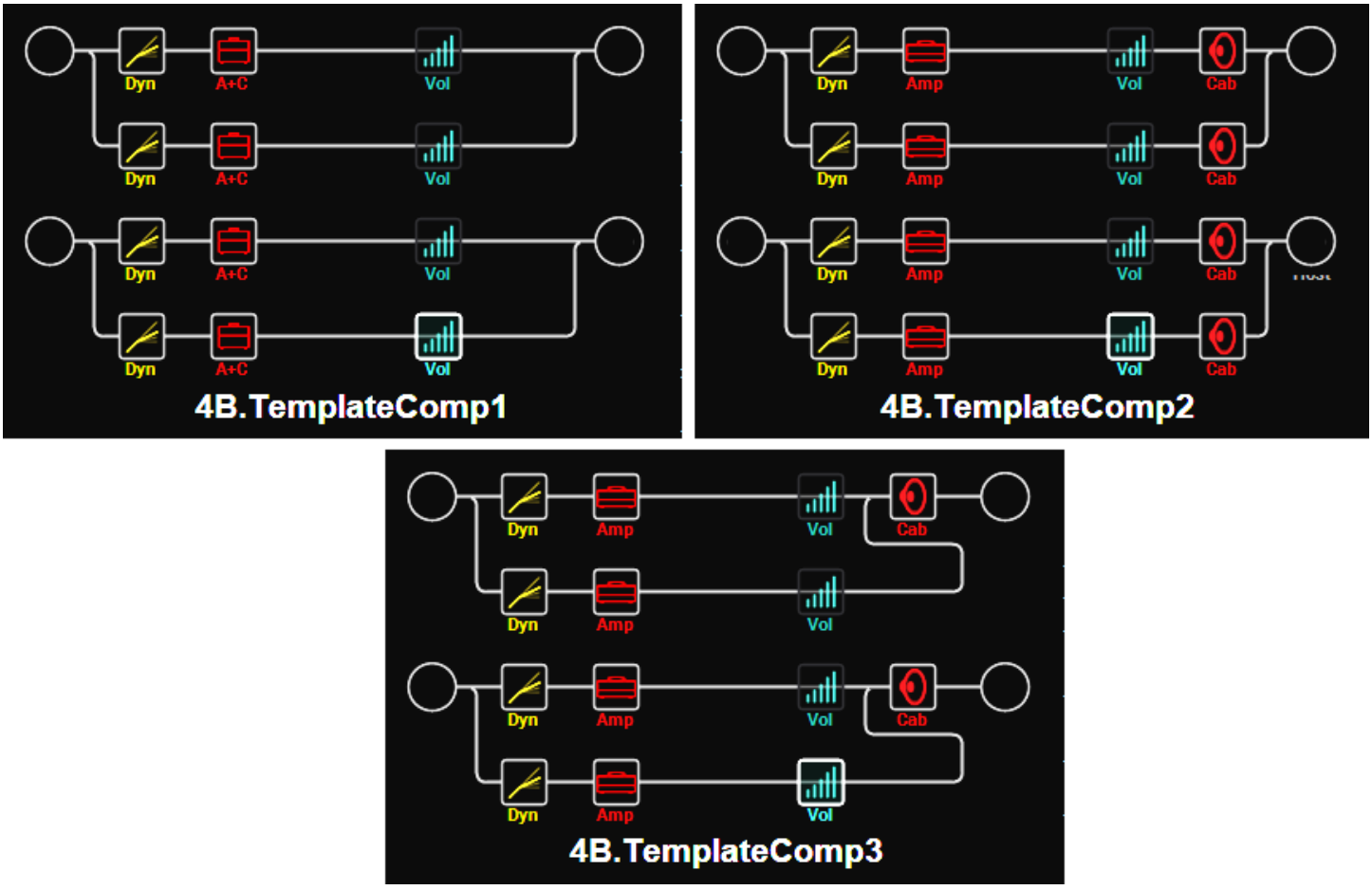


Figure 9.9 4-Band templates based on the 3-Band Comp.

- **4B.TemplateComp1** is the simplest 4-Band template. It inserts an Amp+Cab in each path.
- **4B.TemplateComp2** dedicates a separate amp and cab to each path. This simplifies inserting Cabs that have been saved as Favorites, although you can combine a path's Amp and Cab to use one less block. Note that you can't have Dual Cabs in parallel Paths. The Mixers pan the two midrange bands.
- **4B.TemplateComp3** saves processing power by sharing a Dual Cab between each pair of two amps. These pan full left and right. The Mixers pan the two midrange bands.

4-Band Template Preset that Uses Less Processing Power

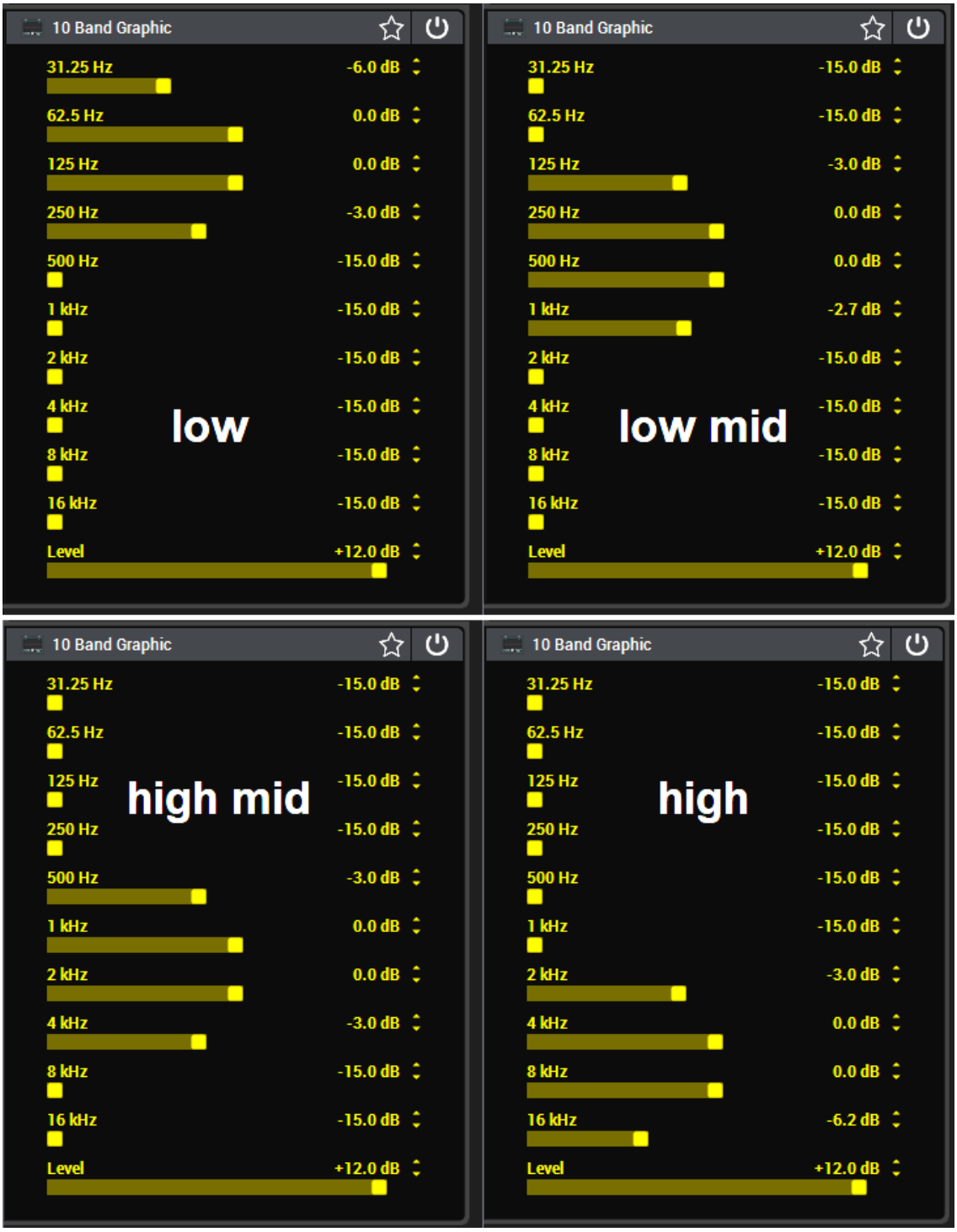


Figure 9.10 10 Band Graphic EQ settings for the four bands.

Splitting with 10 Band Graphic EQs is my usual choice for 4-band splitting. They require less than half the processing power of using 3-Band Comps.

Fig. 9.10 shows the 10 Band Graphic settings for the four bands. Although the bands overlap, these settings give the most even frequency response when summing the bands together. The Favorites folder includes the blocks **4B.LoSplitGEQ.fav**, **4B.LoMidSplitGEQ.fav**, **4B.HiMidSplitGEQ.fav**, **4B.HiSplitGEQ.fav**. These cover the low, low mid, high mid, and high bands respectively. The 10 Band Graphic Level parameters can provide additional control over the overdrive amount for each band's post-EQ effects.

The **4B.TemplateGEQ1**, **4B.TemplateGEQ2**, and **4B.TemplateGEQ3** templates have the same layout as the **4B.TemplateComp1**, **4B.TemplateComp1**, and **4B.TemplateComp3** templates in fig. 9.9. The Low, Low Mid, High Mid, and High 10 Band Graphic blocks replace the equivalent 3-Band Comp blocks.

As with the previous 3-Band templates, remember that you can choose which path handles which frequency range. This is particularly relevant with 4-Band templates, because you might want to process pairs of bands—highs and lows, the two mid bands, the low and lower mid bands, etc. Similarly, using the Vol (Gain) blocks as diagnostic mute buttons simplifies optimizing the sound of individual bands or paths. Then you're not distracted by the sound of the other bands. After creating your preset, delete these to reclaim space for more blocks.

2-Band Template for 4-Path Helix that Requires Minimal DSP Power

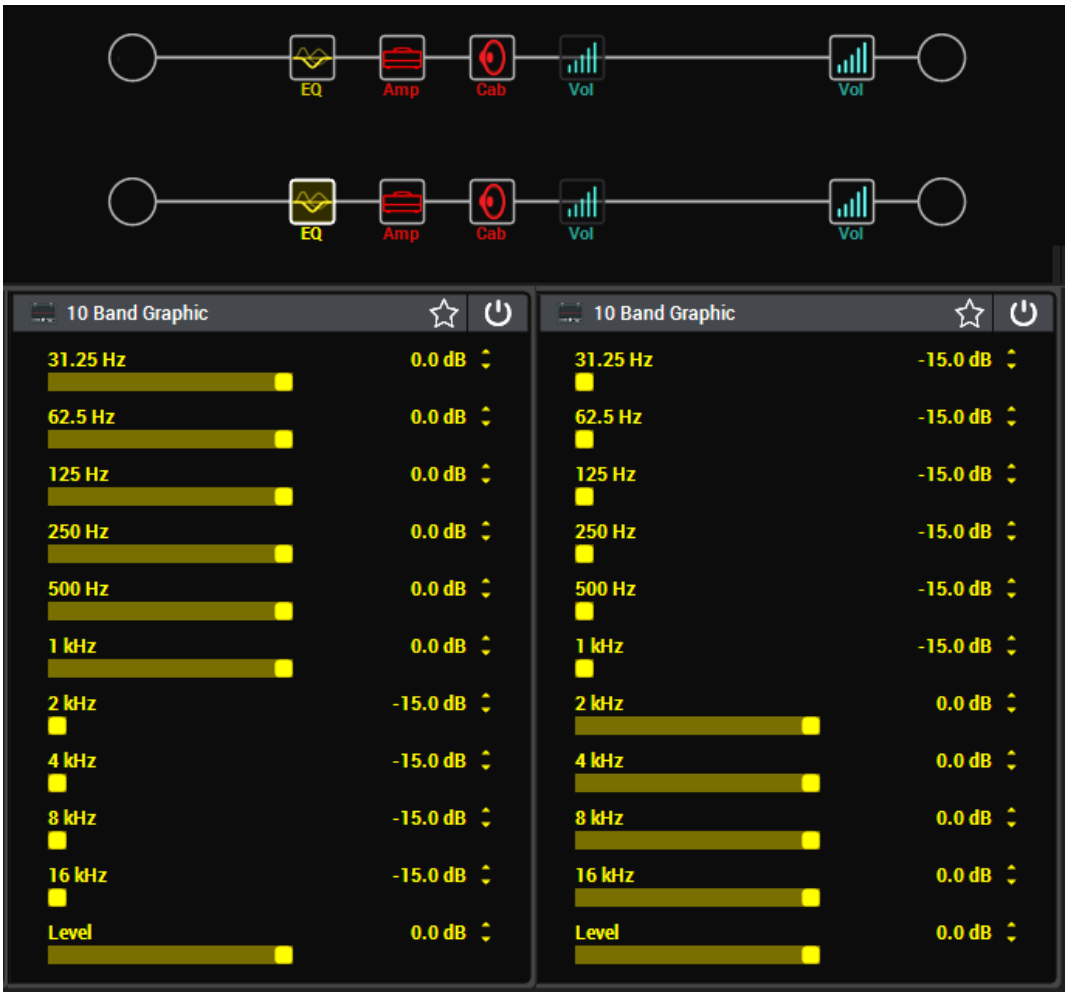


Figure 9.11 The **2B.TemplateGEQ** 2-Band split uses 10 Band Graphic EQs.

These bi-amping options require a 4-Path Helix. So, they aren't included in the Bi-Amp Techniques chapter. Everything in that chapter can work with a 2-Path Helix.

The reason for bi-amping is because some amps require a lot of DSP power. It's not possible to use the Input frequency split to create parallel instances of amps like the PV VitriolCrunch or PV Vitriol Lead in a single path. Each amp needs a dedicated path. Using additional effects with DSP-hungry amps like the Brit Plexi Jump or Mandarin Rocker also requires individual parallel paths.

A 4-Path Helix can isolate one frequency band and amp to Path 1, and another frequency band and amp to Path 2. This also allows each path to use a Dual Cab.

Fig. 9.11 shows the configuration for **2B.TemplateGEQ** (it's also a good WhoWatt 100-based preset). Experiment with the slider settings for different amps. For example, sometimes the 1 kHz slider might sound better turned up in the low band and turned down in the high band, and sometimes vice-versa.

2-Band Template for 4-Path Helix, Requires More DSP Power

3-Band Comp
Gain Reduction
-42 -35 -28 -21 -14 -7 0
Ratio 2:1 3:1 4:1 6:1 8:1 12:1 20:1
Attack 35 ms
Release 200 ms
Lo X Freq 500 Hz
Hi X Freq 20.0 kHz
Level +2.0 dB
Lo Thresh 0.0 dB
Lo Gain +3.5 dB
Mid Thresh 0.0 dB
Mid Gain -60.0 dB
Hi Thresh 0.0 dB
Hi Gain -60.0 dB
3-Band Comp
Gain Reduction
-42 -35 -28 -21 -14 -7 0
Ratio 2:1 3:1 4:1 6:1 8:1 12:1 20:1
Attack 35 ms
Release 200 ms
Lo X Freq 500 Hz
Hi X Freq 20.0 kHz
Level +2.0 dB
Lo Thresh 0.0 dB
Lo Gain -60.0 dB
Mid Thresh 0.0 dB
Mid Gain +4.0 dB
Hi Thresh 0.0 dB
Hi Gain -60.0 dB

Figure 9.12 The **2B.TemplateComp.fav** 2-Band split uses the 3-Band Comp to provide the frequency splits. The sliders outlined in blue show which bands' Gain need to be turned up. All other Gain sliders are at -60.00 dB.

The configuration is identical to fig. 9.11, expect that 3-Band Compressors replace the 10 Band EQs. The compressor's Lo X Freq parameter sets the split point between the high and low bands. Hi X Freq is irrelevant. All Thresholds are set to 0.0 dB. For the low band, turn up Lo Gain, and turn down Mid Gain and High Gain. For the high band, turn up Mid Gain and turn down Low Gain and High Gain.

The compressor block settings in fig. 9.12 are in the Favorites folder as **2B.HiSplitComp.fav** and **2B.LoSplitComp.fav**. The template **2B.TemplateComp.fav** uses Dual Cabs in Paths 1 and 2.

Integrating EQ-Based Cabs

Chapter 6 includes a section on creating EQ-based cabs. Some presets in the *Free Files > Chapter 9 Presets* folder replace one or more cabs with parametric EQ-based cabs. Fig. 9.13 shows a typical configuration. To determine what works best with the EQ-based cabs, the amp and cab in the middle paths are separate to simplify trying out different combinations. They may be combined into an Amp+Cab, or replaced with EQ.

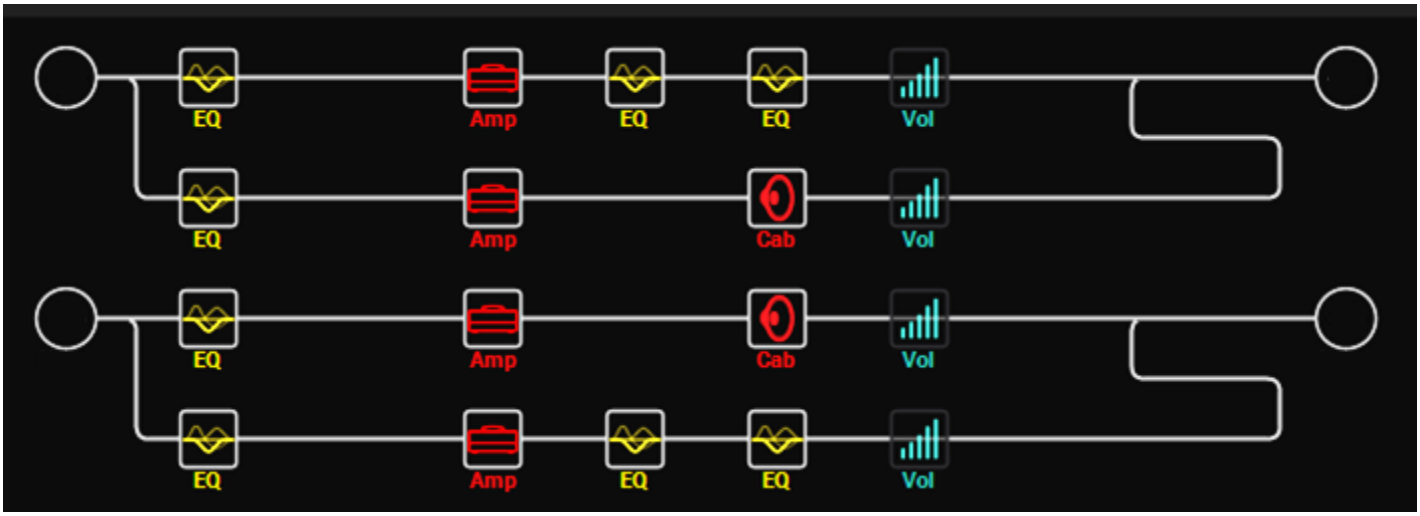


Figure 9.13 Some presets use this configuration, which incorporates EQ-based cabs.

Differences Among the Various Split Methods

The following is subjective, because much depends on the amp itself.

- **3-Band Comp** splits generally give the most focused sound.
- **10 Band Graphic** splits have more overlap. This "spreads" the sound more and is often desirable.
- **Simple EQ** splits have the most overlap. The sound is more like a combination of a single-band and multiband amp.

Practical Examples and Free Presets

Let's load some 3- and 4-band "amp replacement" presets. Although the processing power limits the number of additional effects, you may find the sound is sufficiently satisfying with few, if any, effects because:

- The amp and cab blocks can be different for each band, so you can optimize the bands individually. You might want a big, powerful low-frequency sound, with a cleaner sound for the highs. Try different amps and cabs with different bands.
- Because the two sets of sub-paths feed the Mixers, panning can create stereo imaging. The imaging doesn't rely on effects, so stereo amps collapse perfectly to mono.
- There's less intermodulation distortion. Low, open strings can ring out, even while you play high up on the neck. You'll hear both elements more clearly than with conventional distortion.
- Optimizing Bias, Drive, mic position, and other amp parameters can make high-frequency distortion more defined and articulated.

- Sharing a Single cab between two amps saves processing power and doesn't sound much different compared to using individual cabs for each amp. This is because multiband distortion reduces intermodulation distortion within the amp itself. The cabs don't add distortion, so multiband distortion's main benefits happen before the audio reaches the cabs. For stereo, Dual Cabs draw a bit more DSP power, but are often efficient enough that you can add the effects you need.
- Using more bands is not always better. Sometimes a 3-band setup will give the right sound. Other times, only a 4-band or bi-amp setup will do.
- Using a mix of EQ-based and modeled cabs, or only EQ-based cabs, gives different sound qualities. For more information on creating EQ-based cabs, see Chapter 6. Sometimes I prefer an EQ-based cab's "analog" sound compared to a more authentic tone. Note that unlike Legacy cabs, compared to version 3.50 cabs EQ-based cabs give no significant DSP savings. It's solely about what sound you like.
- Unlike Single Cabs, post-Mixer, stereo EQ-based cabs maintain any panning done in the Mixer.

New 4-Path Amps

The Craig Anderton's Amazing Multiband Helix Presets pack was designed to turn Helix into a "drop-in" amp replacement for 60 guitar amps—one for each amp model at the time. The presets split the guitar into 2 to 4 bands, and process each band individually.

Since then, Helix has added several more amps. Previous eBook editions included presets for the new amps. The presets in this update have been redone from the ground up to take advantage of the version 3.50 cabs. The presets have also been tweaked for consistent perceived volume, and include other enhancements.

As with the original pack, there are no multiband bass amp presets. Multiband techniques generally don't offer the same benefits for bass as with guitar. Bi-amped and parallel bass presets are usually sufficient.

Please note:

- The setlist **1.2 Multiband Amps.hls** includes all the new 4-Path amps for easy comparison.
- The preset's prefix indicates whether it uses **2**, **3**, or **4 B**ands.
- The **dot** between the prefix and name indicates compatibility with only 4-path devices.
- The suffix **E** represents an "eco" preset that conserves DSP power. This allows inserting more effects.
- The suffix **X** means the preset includes an interesting technique, as explained in the description.
- The presets take advantage of the ability to customize each band for the best sound.
- Any presets from previous eBook versions have been updated.
- The amps pair with default cabs, except where noted.
- For adding effects, you're on your own—the presets have varying amounts of leftover DSP power.

RevvGen Red Presets

4B.RevGenRed Each of the four 10 Band Graphic EQ splits drives an Amp+Cab block.
3B.RevGenRed 3-Band Comp blocks provide the splits. Path 2 uses a Dual Cab with a 4x12 Cali V30 and 2x12 Blue Bell Cab. Path 1 has two splits, with Amp+Cab blocks in parallel.
4B.RevGenRedE 10 Band Graphic splits drive two sets of parallel paths, each with a shared cab to save DSP power. Dual Cabs will also work but use more DSP power. The two Vol blocks at the end pan the mono cabs.
4B.RevGenRedX 10 Band Graphic splits drive two sets of parallel paths. Each parallel path replaces one sub-path's cab with an EQ-based cab for a more diffused sound.

2B.RevGenRed 2-band preset uses 10 Band Graphic-based splitting. Each path uses a Dual Cab (4x12 Cali V30 and 1x12 Cali EXT Cab in one path, 4x12 Cali V30 and 2x12 Mail C12Q in the other).
2B.RevGenRedX 2-band preset splits with 10 Band Graphic EQs, but adds IR blocks. Load the Amp Sweetener IR into the IR blocks. Adjust their Levels as needed. The two final Vol blocks pan the mono paths.

RevvGen Purple Presets

4B.RevGenPurple Each of the four 10 Band Graphic splits drives an Amp+Cab block.
3B.RevGenPurple The 3-Band Comp splits drive Amp+Cab blocks. Path 2 uses a Single cab. You could replace this with a Dual Cab, but the preset would require more DSP power.
4B.RevGenPurpleE 10 Band Graphic splits drive two sets of parallel paths, each with a shared cab to save DSP power. You could replace either or both cabs with a Dual Cab, but then the preset would require slightly more DSP power. The two Vol blocks at the end pan the mono cabs.
4B.RevGenPurpleX 10 Band Graphic splits drive two sets of parallel paths. Each of these replaces one sub-path's cab with an EQ-based cab, for a more diffused, "analog" sound.

GrammaticoLG Presets

4B.GramatcLGNrm, **4B.GramatcLGBrt**, **4B.GramatcLGJmp** Each of the four 10 Band Graphic splits drives an Amp+Cab block.
3B.GramatcLGNrm1, **3B.GramatcLGBrt1**, **3B.GramatcLGJmp1** The 3-Band Comp splits drive Amp+Cab blocks. Path 2 uses a single cab. You could replace this with a Dual Cab, but the preset would use more DSP procession power.
3B.GramatcLGNrm2, **3B.GramatcLGBrt2**, **3B.GramatcLGJmp2** Two 3-Band Comp splits drive Path 1, with parallel Amp+Cab blocks for the mid and hi sub-paths. Another 3-Band Comp drives Path 2, with an amp and Dual Cab (1x12 Grammatico and 4x12 Uber T75).
4B.GramatcLGNrmE, **4B.GramatcLGBrtE**, **4B.GramatcLGJmpE** 10 Band Graphic splits drive two sets of parallel paths, each with a shared cab to save DSP power. The two Vol blocks at the end pan the mono cabs.
4B.GramatcLGNrmX, **4B.GramatcLGBrtX** 10 Band Graphic splits drive two sets of parallel paths. Each path replaces one sub-path's cab with an EQ-based cab for a more diffused sound. **4B.GramatcLGJmpX** is similar but replaces all cabs with EQ-based cabs. The stereo imaging varies subtly and dynamically.
2B.GramaticLGNrm 2-band preset uses 3-Band Comp-based splitting, with Dual Cabs in each path (1x12 Grammatico and 4x12 Greenback 25, 1x12 Grammatico and 4x12 Blackback 30).
2B.GramatcLGJmpX 2-band preset uses 10 Band Graphic-based splitting but adds IR blocks. Load the Amp Sweetener IR into the IR blocks. Adjust their Levels as needed. The two Vol blocks at the end pan the mono paths in the stereo field.

Fullerton Presets

4B.FullertonNrm, **4B.FullertonBrt**, **4B.FullertonJmp** Each of the four 10 Band Graphic splits drives an Amp+Cab block.
3B.FullertonNrm, 3B.FullertonJmp The three splits use 10 Band Graphic blocks. Path 1 uses a Dual Cab with two 1x12 Grammatico Cabs. Different mic setups give stereo imaging. Path 2 has two splits, with Amp+Cab blocks in parallel.
3B.FullertonBrt The three splits use 10 Band Graphic blocks. Path 1 (mids) uses a Dual Cab, but with Mandarin EM Cabs instead of the Fullerton's default 1x12 Grammatico Cabs. Different mic setups give a stereo image. Path 2 splits to low and high frequencies, with Amp+Cab blocks in parallel.
4B.FullertonNrmE, **4B.FullertonJmpE** 10 Band Graphic splits drive two sets of parallel paths, each with a shared cab to save DSP power. You could replace either or both cabs with a Dual Cab, but then the preset would require more DSP power. The two Vol blocks at the end pan the mono cabs.

4B.FullertonNrmX 10 Band Graphic splits drive two sets of parallel paths. Each of these replaces one sub-path's cab with an EQ-based cab for a more diffused, "analog" sound.
4B.FullertnBrtE1 Uses 3-Band Comp splits for the low, low mid, and high mid paths, and a 10 Band Graphic split for the high frequencies. The parallel paths share a cab to save DSP power. You could replace either or both cabs with a Dual Cab. The preset would then require more DSP power, but provide a stereo output.

4B.FullertnBrtE2 10 Band Graphics provide the splits. Shared cabs (a 1x12 Grammatico and a 2x12 Mail C12Q) for the parallel paths save DSP power. You could replace either or both cabs with a Dual Cab. The preset would require more DSP power, but provide a stereo output.
4B.FullertonJmpX 10 Band Graphic splits drive two sets of parallel paths. Each of these replaces one sub-path's cab with an EQ-based cab for a more diffused and "fuzzy" sound.

Peavey Presets

The PV VitriolCrunch and PV Vitriol Lead Amps require the most DSP power of any amps. It's not possible to insert them in parallel in a path, let alone do true 4-band processing. So, these presets use some interesting techniques to create multiband presets.

4B.PVitriCrnchX1, 4B.PVitriLeadX1 10 Band Graphic blocks provide the four splits. The low mid and high mid paths use the Amp+Cab, but there's enough DSP power left over to use the VitriolCrunch Preamp for the less-critical low and high paths. The Preamps use the same cab as the amp, and the sound is similar.
2B.PVitriCrnchX2 2-band preset uses 10 Band Graphic-based splitting, and Dual Cabs for each path's amp. The two Vol blocks are hard panned left/right (ideal for LCR mixing). Bypass them for more subtle panning.
4B.PVitriCrnchX3 This 4-band variation uses 10 Band Graphic-based splits, but is intended not to sound like the "real" amp—the cutting, dense, buzzy tone isn't the normal physical-world sound. Preamps with cabs cover the high and low-frequency paths. The low mid and high mid paths use an amp, followed by a single stage of parametric EQ to provide the cab.
2B.PVitriLeadX2 This is like **2B.PVitriCrnchX2**, except that it uses a Single cab instead of a Dual Cab. This frees up DSP power for effects: in each path, you can add an LA Studio Comp and Tone Sovereign Distortion between the 10 Band Graphic frequency split and Amp+Cab, *and* a stereo 4-Voice Chorus and Stereo Dual Delay after the Amp+Cab. Or, choose mono output effects and have more options. In that case, the panpot blocks at the output of each Path are handy for creating a stereo image.
4B.PVitriLeadX3 The configuration is the same as **4B.PVitriCrnchX3**, but the elements are optimized for a brighter, cutting lead sound.

Line 6 Ventoux Presets

4B.Line6Ventoux Each of the four 10 Band Graphic splits drives an Amp+Cab block.
3B.Line6Ventoux The 3-Band Comp splits drive Amp+Cab blocks. Path 2 uses a Single cab. You could replace this with a Dual Cab, but the preset would require more DSP power.
4B.Line6VentouxE 10 Band Graphic splits use shared Dual Cabs (2x12 Mail C12Q and 4x10 Tweed P10R, 2x12 Mail C12Q and 2x12 Blue Bell). Save more DSP by replacing the Dual Cabs with Single cabs.
2B.Line6VentouX1 This 2-band preset extracts a different kind of sound from the Ventoux. It uses 10 Band Graphic-based splitting but adds IR blocks. Load the Amp Sweetener IR into the IR blocks. Adjust their Levels as needed. The two Vol blocks at the end pan the mono paths.
2B.Line6VentouX2 3-Band Comp-based splitting for the two bands gives a focused sound. There's a big, rough bottom, and a "chimey" brightness on top. The IR blocks are optional. Load the Amp Sweetener IR into them, and adjust their Levels as needed. The two Vol blocks at the end pan the mono paths.
4B.Line6VentouX3 This uses 10 Band Graphics to do the splits, and a single Parametric EQ stage as the "cab" for each amp. The result is a big, sprawling sound that's drenched in fuzz.

Benzin/Das Benzin Presets

4B.Benzin Mega1 Each of the four 3-Band Comp-based splits drives an Amp+Cab block.
4B.Benzin Mega2 This is like **4B.Benzin Mega1** but shows how changing mic positions in the cabs—even with the same mics—alters the sound.
4B.Benzin Mega3 This is like **4B.Benzin Mega1** but uses 10 Band Graphic-based splits.
3B.Benzin Mega The 3-Band Comp splits drive Amp+Cab blocks. Path 2 uses a Single cab. You could replace this with a Dual Cab, but the preset would use more DSP power.
4B.Benzin MegaE Each of the four 10 Band Graphic-based splits drives an amp block. The two sets of parallel paths share Dual Cabs to conserve DSP power. Each Dual Cab has the same speakers (4x12 Cali V30 and 4x12 Greenback 25), but the mic placement parameters are quite different.
4B.DasBenznLead Each of the four 3-Band Comp-based splits drives an Amp+Cab block.
3B.DasBenznLead The 3-Band Comp splits drive Amp+Cab blocks. Path 2 uses a Single cab. You could replace this with a Dual Cab, but the preset would require more DSP power.
4B.DasBenznLeadE Each of the four 10 Band Graphic-based splits drives an amp block. The two sets of parallel paths share Single Cabs to conserve DSP power. The two Vol blocks at the end pan the mono paths.
4B.DasBenznLeadX The 10 Band Graphic splits drive amps. Each amp is followed by a pseudo-cab made up of two Parametric EQ blocks, The sound is different from the other Das Benzin Lead sounds.

Moo)))n Presets

4B.Moo)))n Nrm, **4B.Moo)))n Brt**, **4B.Moo)))n Jmp** Each of the four 10 Band Graphic-based splits drives an Amp+Cab block.
4B.Moo)))n NrmE 10 Band Graphic splits drive two sets of parallel paths, each with a shared Single cab to save DSP power. You could replace either or both cabs with a Dual Cab, but the preset would require more DSP power. The two Vol blocks at the end pan the mono cabs.
4B.Moo)))n Nrm X1 This preset demonstrates how using different cabs in each path obtains a different sound compared to using the same cab in all paths. Each of the four 10 Band Graphic splits drives an amp block. Each amp block is followed by its own cab.
4B.Moo)))n Nrm X2 10 Band Graphic splits drive two sets of parallel paths. Each parallel path replaces one sub-path's cab with an EQ-based cab for a more diffused sound.
4B.Moo)))n JmpX I had so much fun using different cabs in **4B.Moo)))n Nrm X1** that this preset follows the same concept.

US Princess Presets

The four US Princess presets are about variations on a theme.

4B.US Princess1 Each of the four 10 Band Graphic-based splits drives an Amp+Cab block.
4B.US Princess2 3-Band Comp splits create a cleaner version of **4B.US Princess1**.
3B.US Princess This uses the Simple EQ for the three splits. I wanted a preset with a big, fat, stereo low end, and the Simple EQ's leakage between its high, low, and mid bands was the perfect way to drive a Dual Cab with a 4x12 Blackback 30 and 4x12 Greenback 25.
4B.US PrincessE 10 Band Graphic splits drive two sets of parallel paths, each with a shared Single cab to save DSP power. You could replace either or both cabs with a Dual Cab, but the preset would require more DSP power. The two Vol blocks at the end pan the mono cabs.
4B.US PrincessX Like **4B.US Princess1**, four 10 Band Graphic-based splits drive an Amp+Cab block. However, the low and low mid paths use 4x12 Blackback 30 Cabs, while the high mid and high paths use 1x10 US Princess Cabs. This preset is excellent for LCR mixing.

Mandarin Rocker Presets

4B.MandrnRockr Each of the four 10 Band Graphic-based splits drives an Amp+Cab block.
3B.MandrnRockr1 Each of the 3-Band Comp-based splits drives an Amp+Cab block.
3B.MandrnRockr2 This is like **3B.MandrnRockr1**, but with a darker, beefier sound.
4B.MandrnRockrE1 This bright, jangly preset drives four amps with 3-Band Comp splits. Paths 1 and 2 have parallel sub-paths that each share a Dual Cab. Replace the Dual Cabs with Single cabs for more DSP power.
4B.MandrnRockrE2 The goal for this preset is a more "midrangy" sound. 10 Band Graphic splits drive four amps. Paths 1 and 2 have parallel sub-paths that each share a Dual Cab. Replace the Dual Cabs with Single cabs to free up DSP power.
4B.MandrnRockrE3 3-Band Comp splits drive four amps for a focused sound. Paths 1 and 2 mix into Single cabs. You could replace either or both cabs with a Dual Cab, but the preset would require more DSP power. The two Vol blocks at the end pan the mono cabs.
4B.MandrnRockrE4 10 Band Graphic splits drive four amps for a focused sound. Paths 1 and 2 mix into Single cabs. You could replace either or both cabs with a Dual Cab, but the preset would require more DSP power. The two Vol blocks at the end pan the mono cabs.
4B.MandrnRockrX1 10 Band Graphic splits drive two sets of parallel paths. The low- and high-frequency paths use EQ-based cabs for a more diffused, "analog"-type sound.
2B. MandrnRockrX2 10 Band Graphic EQs provide the splits. The IR blocks are optional. Load the Amp Sweetener IR into them, and adjust their Levels as needed. The two Vol blocks at the end pan the mono paths.

3-Band Amps with Effects

3-Band presets free up processing power for adding effects. Inserting them in the separate path (i.e., not the two parallel sub-paths) gives more potential processing power than inserting them in the sub-paths. Effects are the key to creating multiband lead presets.

This section has a DIY orientation, because what guitarists want from lead presets varies. We'll describe which tradeoffs maximize processing power, and what's possible with three bands. Hopefully, you'll be inspired to come up with your own signature sounds. Here are some tips about creating 3-Band lead presets:

- The midrange band usually goes in Path 1. The guitar's midrange dominates, so it benefits the most from multiple effects.
- The low and high bands operate in parallel, typically in Path 2. Depending on the amps, and whether they share a cab or use EQ-based cabs, there can be enough processing power for some effects.
- You can mix and match the blocks that split the audio into bands. For example, splitting the low and high bands with the Simple EQ saves processing power compared to the 10 Band Graphic or 3-Band Comp-based splitting options. Then, you can use the 3-Band Comp midrange block for the midrange path. This gives the most focus to the midrange band.
- Explore different tradeoffs. For example, low- and high-band amps that don't need as much DSP power may nonetheless do what's needed, which would allow wrapping more effects around them. Processing the midrange might be more critical to obtaining the desired sound.
- 3-Band presets offer unique stereo possibilities for the studio. For live use, you may want to stay with mono. Because mono effects need less processing power than stereo ones, your live performance presets can accommodate more effects, or amps that need more processing power.

3B.Echoing LD Preset

Fig. 9.14 shows **3B.Echoing LD**, a typical 3-Band preset for leads. Here's what's inserted in the various paths:

- **Path 1 (Midrange):** Deluxe Comp, Simple EQ for Mid split, Scream 808 Overdrive, Tweed Blues Nrm Amp, 4x10 Tweed P10R Cab, Stereo Cosmos Echo, Stereo Dynamic Hall reverb, Stereo Chorus.
- **Path 2 (Low and High in parallel):** Alpaca Rouge distortion, LA Studio Comp, Split Y.
- **Low Sub-Path:** 10 Band Graphic for Low split, Who Watt 100 Amp, Harmonic Tremolo, 4x12 Uber V30 cab.
- **High Sub-Path:** 10 Band Graphic for Hi split, Jazz Rivet 120 Amp, Harmonic Tremolo, 1x10 US Princess Cab.
- **Post-Mixer Output:** Stereo Simple EQ for shaping tone, stereo Simple Delay for slapback echo.

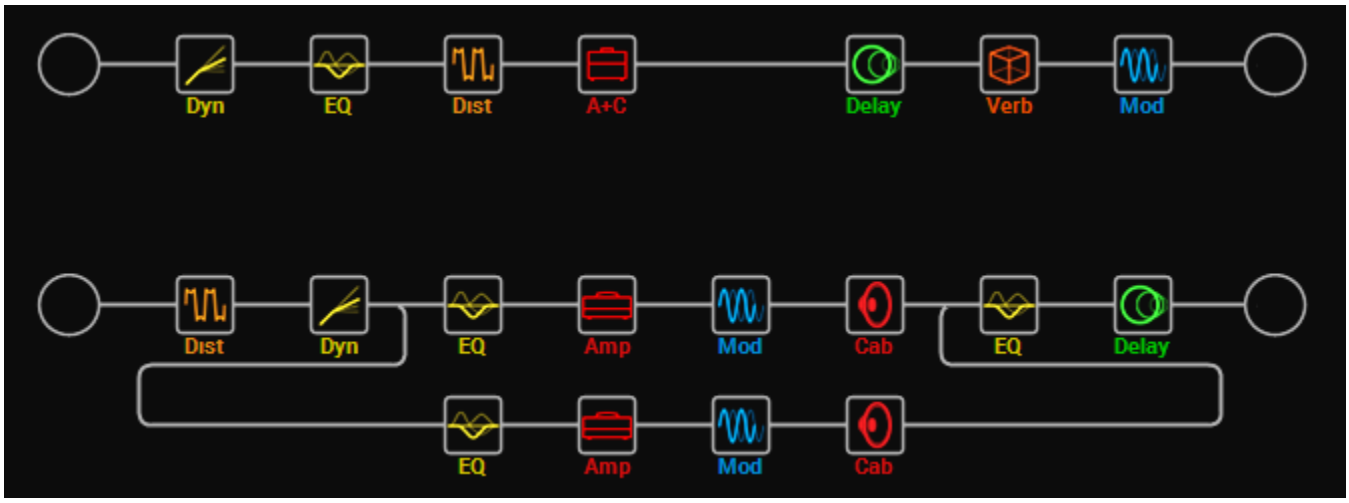


Figure 9.14 The **3B.Echoing LD** preset takes advantage of a 3-Band preset's additional processing power.

3-Band FullertnNrmLD Preset

The **3B.FullertnNrmLD** preset (fig. 9.15) has several interesting twists:

- **Path 1 (Midrange):** 3-Band Comp for Mid split, Fullerton Nrm Amp, Autofilter, Alpaca Rouge distortion, Dual Cab (two 1x12 Grammatico Cabs with different mic placement for stereo), stereo Dynamic Hall reverb, stereo Dual Delay, Simple EQ for shaping tone.
- To add some midrange compression, the Mid band's 3-Band Comp lowers the mid's Threshold.
- Placing the Autofilter after the amp, but before Distortion, gives motion to the sound. You'll need to edit the Autofilter Sensitivity parameter to match your playing style.
- **Path 2 (Low and High in parallel):** Split Y at input. Mixer pans the two paths at the output.
- **Low Sub-Path:** 3-Band Comp for Low split, Fullerton Nrm Amp, 1x10 US Princess Cab.
- **High Sub-Path:** 3-Band Comp for High split, Fullerton Nrm Amp, 4x12 1960A T75 Cab.
- **Path 2** has enough DSP left over to insert one of several distortions prior to the split, or one of several stereo effects at the output.

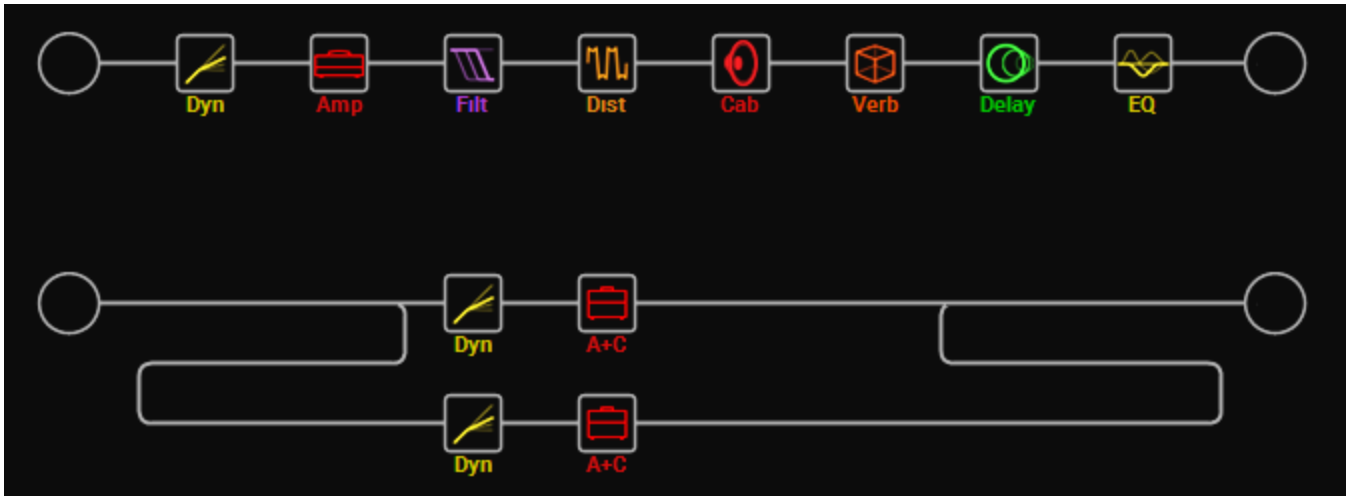


Figure 9.15 The 3-Band **FullertnNrmLD** preset fills the midrange path with effects.

3-Band FulertnNrmLD2 Preset

This variation (fig. 9.16) on the **3B.FullertnNrmLD** preset keeps Path 1 the same but shares a Dual Cab for Path 2. The settings are the same as for the Dual Cab in Path 1, except that the pan positions for the two speakers are reversed.

Path 2 "frames" the mids with the High and Low paths. This has less of a stereo effect than **3B.FullertnNrmLD**. Removing the stereo Simple Delay at Path 2's output frees up the DSP needed to insert one of several distortions prior to the split.

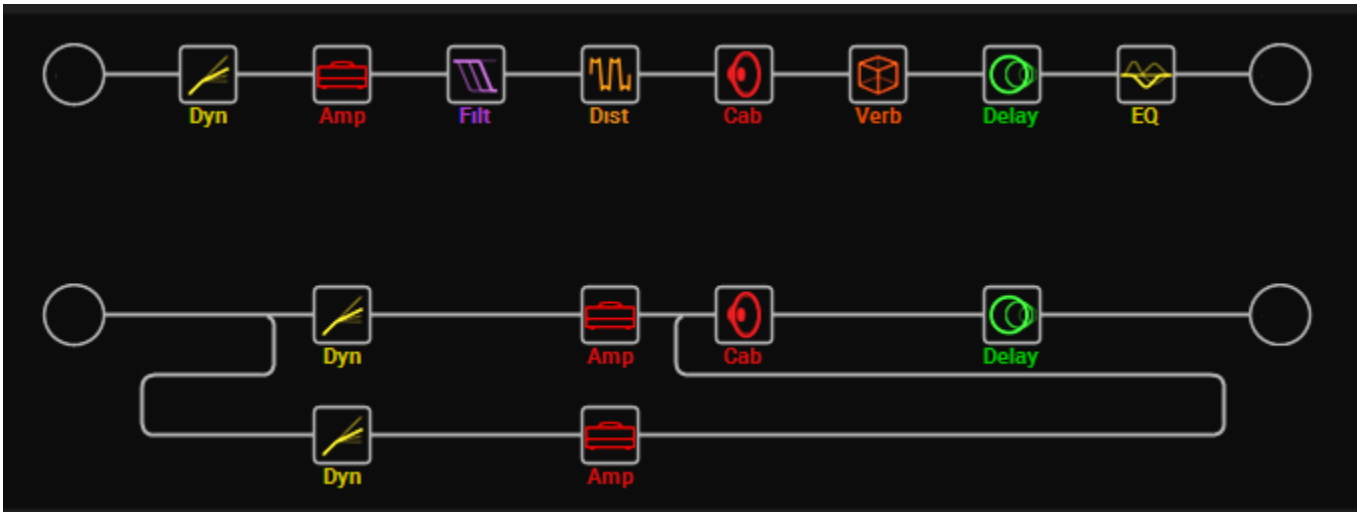


Figure 9.16 Compared to **FullertnNrmLD**, this variation gives a fuller, more mono spread.

3B.MandrinRockLD Preset

A funny thing happened on the way to v1.2 of this eBook. I'm a Mandarin Rocker Amp fan, but before the v3.50 Helix cabs, its hunger for DSP prevented having three amps, cabs, *and* the processing necessary to split the audio into three separate bands. So, Path 2 in the **3B.MandrinRockLD** preset (fig. 9.17) used EQ-based cabs for the low and high splits.

Yet when I replaced the EQ-based cabs with version 3.50 cabs, I liked the original sound better. I'm not that surprised, because I tweaked those EQ carefully to complement the sound from Path 1. However, I did update the cab in Path 1, the output effects to stereo, and tweaked other parameters.

- **Path 1 (Midrange):** 3-Band Comp for Mid split, Minotaur distortion, Vetta Wah, Mandarin Rocker Amp, 4x12 Mandarin EM Cab, stereo 4-Voice Chorus, stereo Double Tank reverb, and stereo Simple Delay to create a stereo image. There's enough DSP to replace the Single cab with a Dual Cab.
- **Path 2 (Low and High in parallel):** Split Y at input.
- **Low Sub-Path:** Simple EQ for the Low split, Low and High Cut filter (conditions the sound going into the amp by reducing the highs), Mandarin Rocker, single-stage Parametric EQ-based cab.
- **High Sub-Path:** Simple EQ for the High split, Mandarin Rocker, two-stage Parametric EQ-based cab.
- **Post-Mixer Output:** Stereo Simple Delay for echo effects.

In **Path 1**, this preset can have significant tonal changes by substituting a different cab, amp, or both. For example, the 1x12 Grammatico Cab gives a grittier, more cutting sound, while the 2x12 Blue Bell focuses more on the midrange.

Path 2 has virtually no DSP power left over. But because of the version 3.50 cabs' efficiency, you can use them to replace the EQ-based cabs. This frees up enough DSP to add a distortion effect (instead of dynamics) prior to the split. Please note that the choice of amps will be limited.

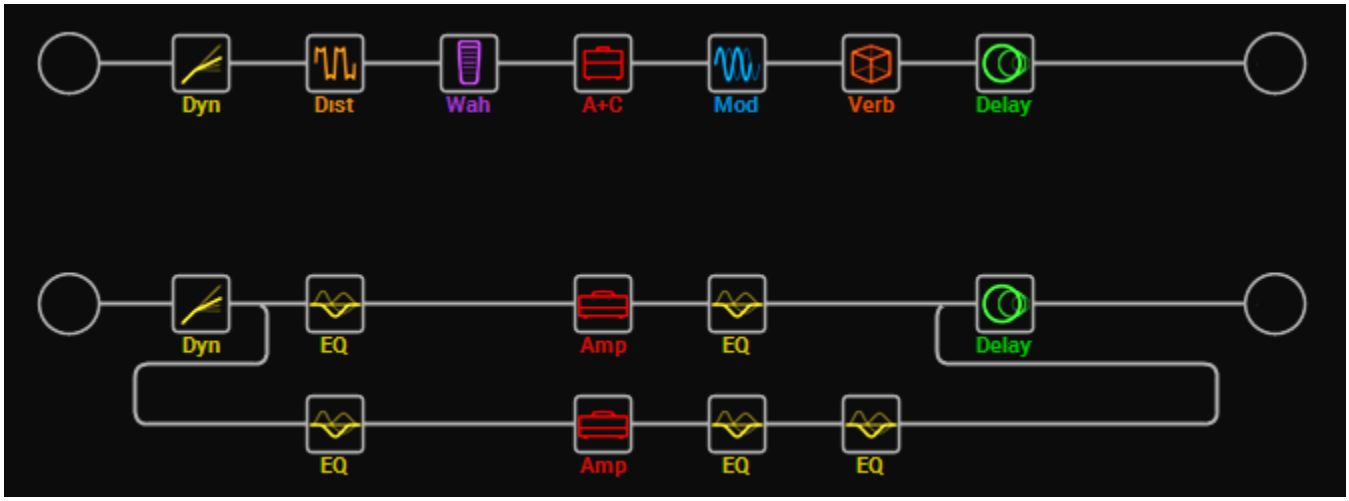


Figure 9.17 The lead preset **3B.MandrinRockLD** shows that EQ-based cabs have their place for lead sounds.

3B.Octav+Verb LD Preset

The **3B.Octav+Verb LD** preset (fig. 9.18) showcases more novel possibilities from splitting the guitar into multiple bands.

- **Path 1 (Midrange):** 3-Band Comp for Mid split, Teemah! distortion, A30 Fawn Brt Amp, Poly Capo mixes straight and octave below, Dual Cab (4x12 MOO)))N T74 and 2x12 Blue Bell), and stereo Simple Delay create a stereo image. The Poly Capo placement is non-traditional (it usually goes at a chain's beginning), but sounds "right." Try placing it before the amp and after the cab. See which you prefer.
- **Path 2 (Low and High in parallel):** The Deluxe Comp prior to input Split Y increases sustain.

- **Low Sub-Path:** 10 Band Graphic for the Low split, Amp+Cab (Mandarin Rocker and 4x12 Mandarin EM). Panned hard left in the mixer.
- **High Sub-Path:** 10 Band Graphic for the High split, Amp+Cab (Mandarin Rocker and 4x12 Mandarin EM). Panned hard right in the mixer.
- **Post-Mixer Output:** Stereo Plateaux reverb.

The midrange path's Poly Capo transposes down an octave, and mixes dry and octave-shifted guitar. If you want to include a distortion block, that limits the amp choices because the Poly Capo needs a lot of processing power. The A30 Fawn Brt works well.

The low and high paths go through their own Amp+Cabs. The subsequent reverb doesn't affect the octave-lower sound in Path 1, so the reverb "floats" above the lead. It's an unusual effect.

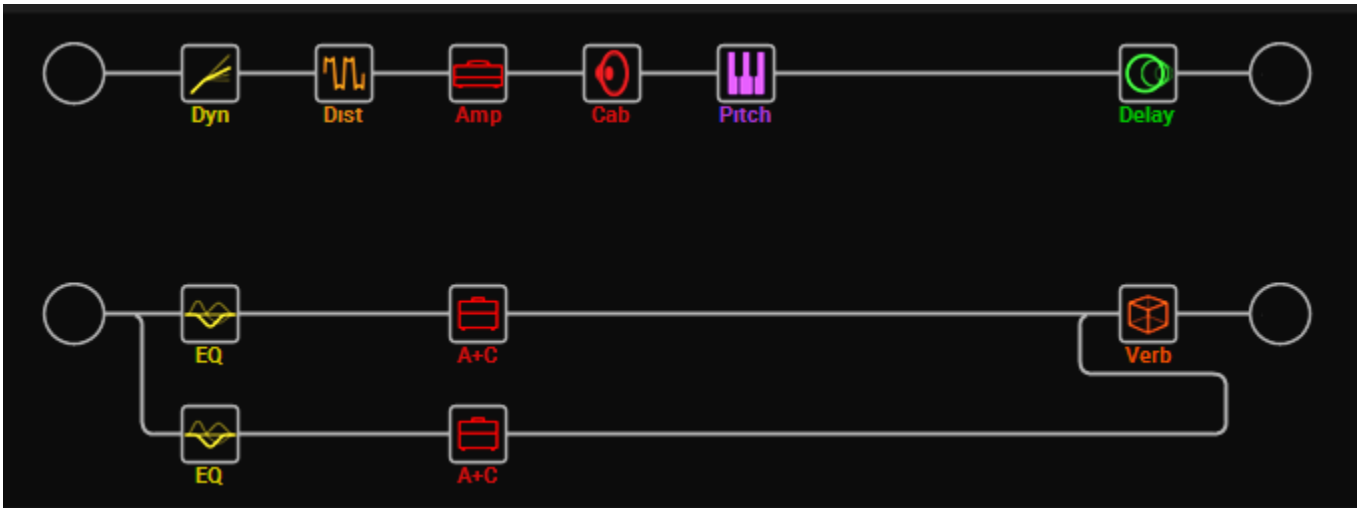


Figure 9.18 This preset produces an unusual combination of octave division and reverb.

3B.+OctaveUp Preset

This preset goes in the other direction. **3B.+OctaveUp** (fig. 9.19) uses three Wringer distortions, which deliver good octave-above effects when playing in higher note ranges (particularly with the neck pickup). The Wringers are the main feature of all three paths.

- **Path 1 (Midrange):** 10 Band Graphic provides a modified Mid split, Wringer distortion, Tweed Blues Brt Amp, 1x12 Grammatico, and Simple Delay.
- **Path 2 (Low and High in parallel):** There are no effects prior to the input Split Y.
- **Low Sub-Path:** 10 Band Graphic provides a modified Low Split that emphasizes the lower mid frequencies. That's followed by a Wringer distortion, and US Small Tweed with 1x12 Grammatico Cab. The output is panned hard left in the Mixer.
- **High Sub-Path:** 10 Band Graphic is a nominal High split that's modified to emphasize the midrange. It's followed by a Wringer distortion, and Tweed Blues Nrm with 1x12 Grammatico Cab. The output is panned hard right in the Mixer.
- **Post-Mixer Output:** The stereo Simple Delay provides a secondary echo effect.

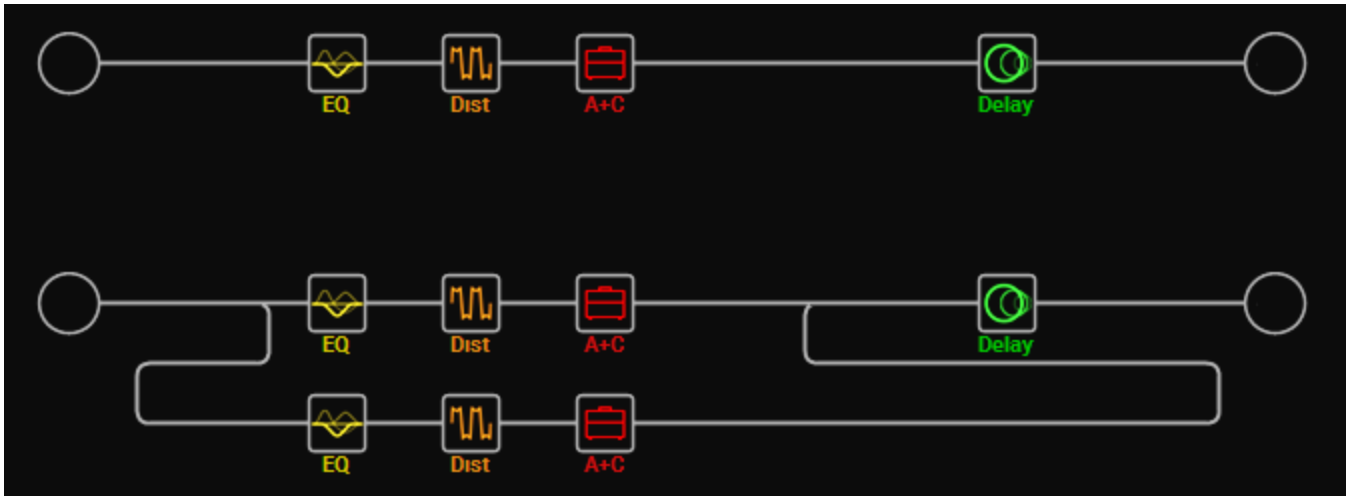


Figure 9.19 Although optimized to give a big stereo sound in the studio, this preset works well in mono.

Customizing the 3- and 4-Band Multiband Presets

- **Important:** Remember that you can use any band for any split. This matters for parallel sub-paths, particularly if they share input and output effects. For example, you may want to feed a shared reverb with the High Mid and High bands, or the Low Mid and Low bands.
- For amp replacement applications, where the goal is a more articulated version of an existing amp, insert the same Amp+Cab in each of the paths. But the possibilities are endless—like a Marshall-type sound for the low bands, and a chimey Vox sound for the high ones.
- Because the signal splits into several paths, each amp gets less signal than usual. The blocks that provide splits for the presets often increase the output level to compensate. However, you may need to turn up an amp's Drive parameter for the desired drive amount.
- Turning down split output levels, or the Drive parameter, is also useful. Reducing drive for the high-frequency path can give a sweet, less harsh sound. Pulling down all the Drives can produce beautiful clean tones.
- You may need to turn down the amps' Master parameter, because the output from four amps adds up.

Snapshots and Multiband Presets

Snapshots tend to fill personal needs, so there are no snapshots in the presets. The Mixer section can be useful for creating multiband preset Snapshots—like no lower band, no middle band, no upper band, bands at different levels, bands panned differently, and so on (fig. 9.20).

To create Snapshot settings for the Mixer, right-click on it. Select the parameter, then choose *Controller Assign > Snapshots.* Repeat this procedure as needed for the other Mixer parameters.

Once you've assigned each parameter you want to change as a Snapshot controller, the assignment remains in effect for all 8 preset Snapshots (or 3 Snapshots for HX Stomp). Select the Mixer, and you can vary the assigned parameters individually for each Snapshot.

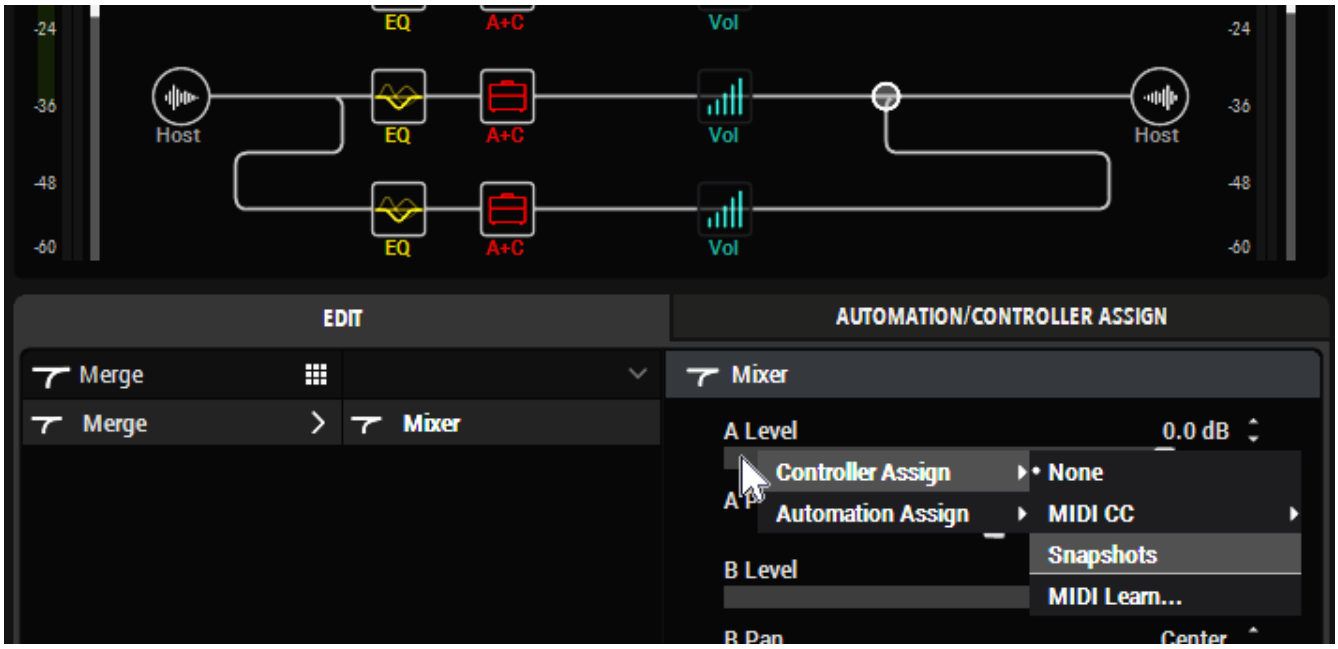


Figure 9.20 The Mixer block is particularly useful for multiband preset Snapshots.

Stereo Possibilities

Most Helix effects are stereo-friendly. Panning in the Mixer section and Dual Cabs can take advantage of them. With a 4-Band preset, spreading the four bands from left to right across the stereo field is one option. For a more centered guitar sound, I prefer to pan the lowest and highest bands to center, and then spread the two middle bands somewhat left and right of center. The Mixer's panning options are also great for those into LCR (left-center-right) mixing.

These multiband presets incorporate stereo capabilities, yet they also collapse well to mono. So, multiband presets can make the transition from studio to stage, and back again.

Surround Sound and Multi-Channel Audio

Immersive audio and surround sound are hot topics. It's uncertain how soon, or even whether, new formats will get traction with consumers. However, Helix is ready.

Helix Floor and Helix Rack can accommodate up to 8 channels of multi-channel audio. A basic application is 4-channel surround sound. Each of the four paths can insert a Send block to direct the path's audio to an individual physical output. For live use, these outputs would feed into a mixer that's part of the surround audio system. In the studio, they would feed separate audio interface inputs. Turning down the Send's Dry Thru prevents audio from going beyond the Send block, if that's needed.

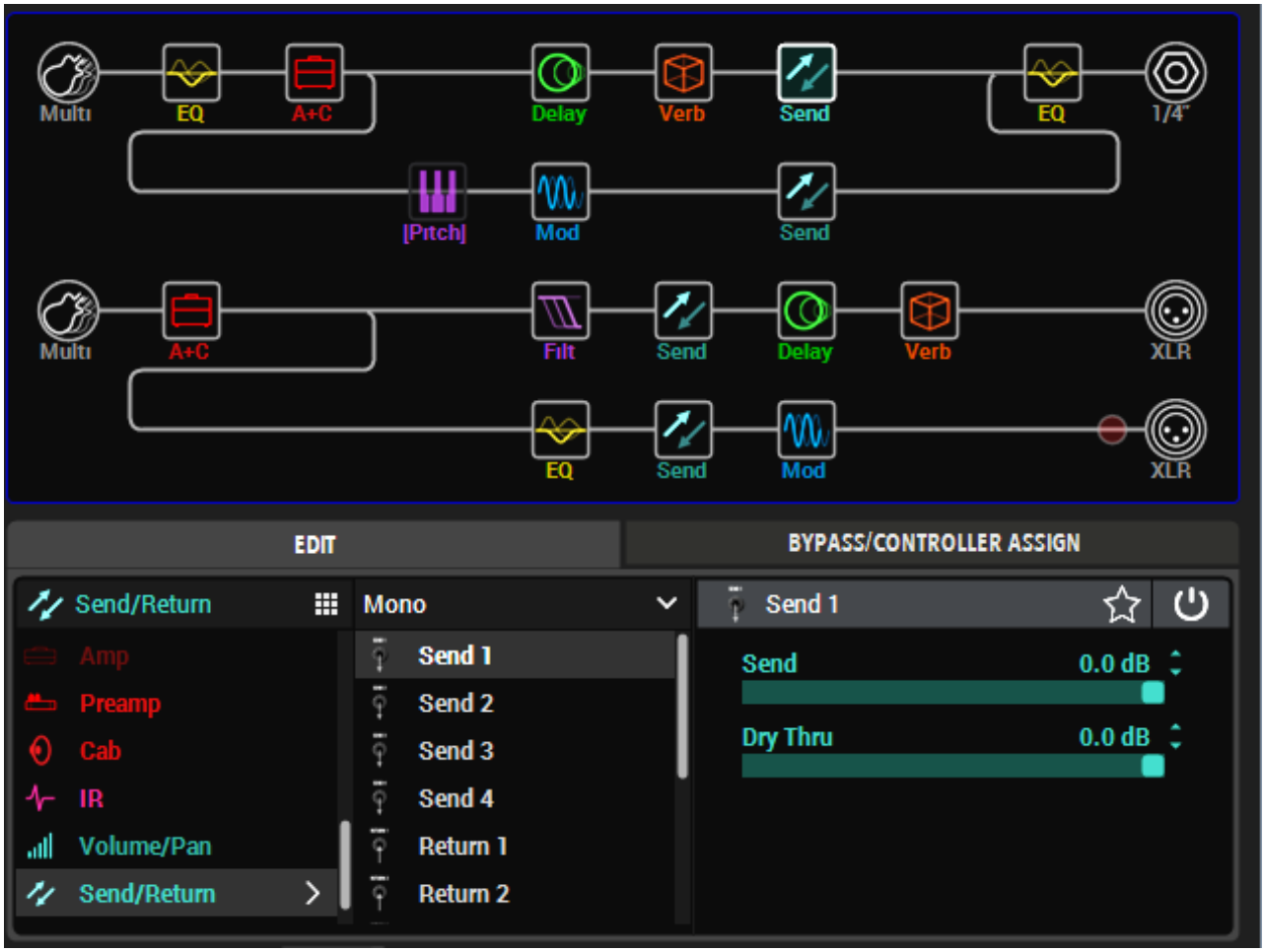


Figure 9.21 8-channel setup for Helix Floor—four channels from sends, and four from two stereo outputs.

Figure 9.21 shows an 8-channel setup. The four Send blocks send individual paths to the four Send output jacks. The Send blocks have Dry Thru turned up so that the audio can also go to the main outputs.

Path 1's post-Mixer EQ isolates the bass frequencies. These go to the main ¼" stereo output and provide a pseudo-LFE (Low-Frequency Effects) channel. A surround system's LFE channel typically feeds a sub-woofer and is band-limited to frequencies below 150-200 Hz or so. It's usually mono, but Helix's output makes two channels available.

Path 2 picks off audio partway through paths 2A and 2B for two Send output jacks. With Dry Thru at 0.0 dB, each path then goes through effects to the XLR outputs. One path pans to the right XLR out, and the other to the left XLR out. This completes the 8-channel audio setup.

Using eight channels requires careful planning when designing a preset. 4- and 6-channel setups are simpler, but the bottom line is Helix has the tools needed for multi-channel audio.

Surround and Multi-Channel Audio with Helix Native

Inserting a Helix Native in multiple buses can produce any number of audio channels, limited only by the computing power needed to run multiple Helix instances. Fig. 9.22 shows a 4.1 surround setup.

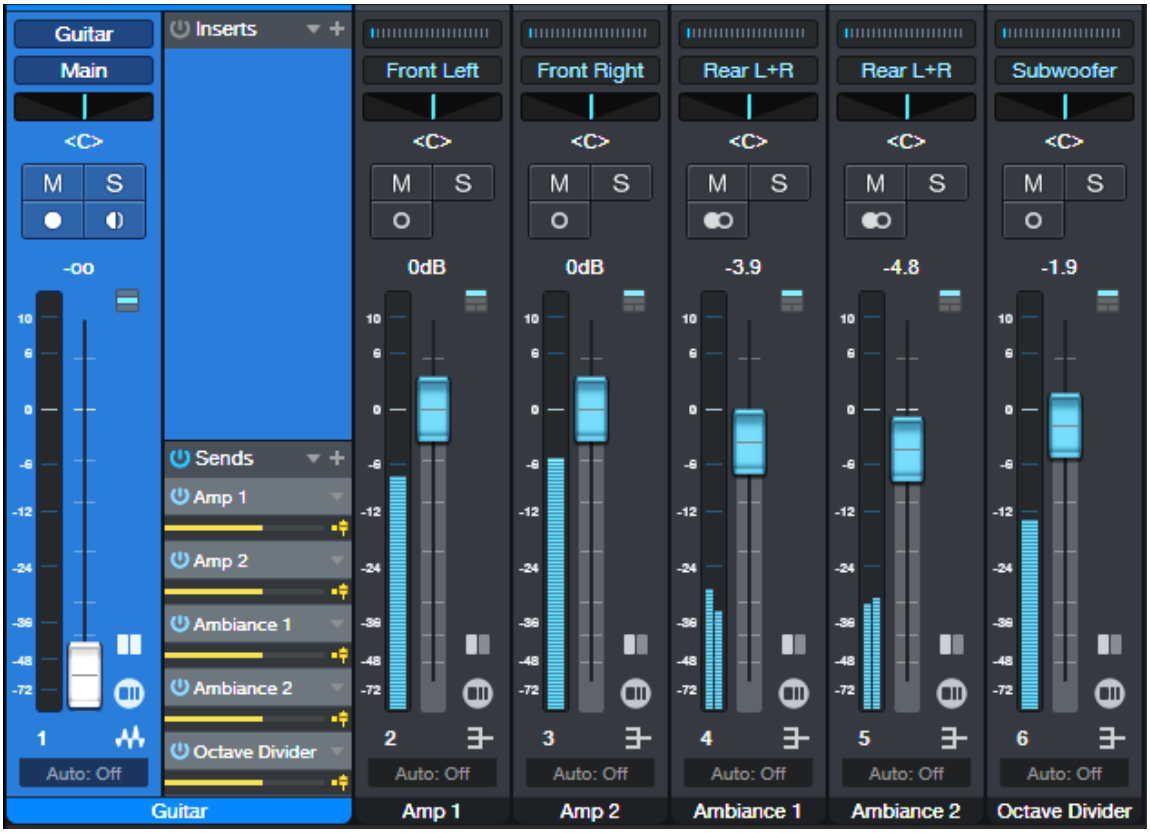


Figure 9.22 4.1 surround configuration, with four channels and a sub-woofer.

The guitar track has five pre-fader sends that go to five buses. Each bus has a Helix inserted. Two buses have amps inserted. With two other buses, Helix is doing ambiance processing with reverb, modulation, and delay. These feed the rear left+right channels in stereo. The Helix in the Subwoofer channel has an octave divider inserted to provide Helix's version of sub-bass.

Chapter 10 | HELIX BASS PROCESSING

Helix is best-known for being a guitar multi-effects, but it also has the amps, cabinets, and effects needed for bass—from simple to sophisticated.

General Tips for Bass

Bass processing has not been explored as much as guitar. Bass is not only a different instrument compared to guitar, but bass needs to fulfill different musical needs:

- Hold down the low end.
- Partner with the drums to make a cohesive rhythm section.
- Maintain a percussive punch in some situations, but be unobtrusive in others.
- Be supportive without being uninteresting.

Notwithstanding effects designed for bass, many effects seem unsuitable for anything other than guitar. Sometimes they aren't "voiced" for bass, and sometimes the effects aren't designed to retain the all-important low end. Of course, there are exceptions, like bassists who use chorusing and delay. However, it's crucial that any effects do not dilute what the bass contributes to an arrangement or mix.

Helix has the expected effects for bass—amps, cabs, dynamics processing, and EQ. However, there are many additional creative options:

- Although effects like distortion are often voiced for guitar, you can take advantage of Helix's parallel signal paths. Dedicate one path to retaining a full, rich bass tone, then layer effects in the other path(s).
- Assigning effect levels to an Expression Pedal allows bringing them in and out strategically. Then, they can supplement what's happening in the bass's main audio path.
- Many effects have sync-to-tempo, so processing can lock to the rhythm. With bass being the bridge between the rhythm section and melodic elements, rhythmic processing enhances that bridge.

Bass presets, whose most common processors are EQ and dynamics, tend to be less complex than guitar presets. Other processors often benefit from being in a parallel path with the main bass sound. Some effects include an associated mix control that can create a blend of dry and processed sound. Although some of the following presets are designed to make cool, flashy sounds, most are designed to be utilitarian and supportive of the bass's crucial role.

Please note:

- 2- and 4-Path compatible, bass-specific files in the *Free Files* > *Chapter 10* > *Presets* folder start with the prefix **B_**
- 4-Path compatible, bass-specific files in the *Free Files* > *Chapter 10* > *Presets* folder start with the prefix **B.**

Main Squeeze 3-Band Comp

The 3-Band Comp splits the signal into three frequency bands (low, mid, and high), and compresses each one individually. However, the default frequency ranges are optimized for guitar, not bass:

- The Lo X frequency (the dividing line between the low and high bands) is 400 Hz.
- The Hi X frequency (the dividing line between the mid and high bands) is 3.0 kHz.
- The default 35 ms attack time isn't long enough for the bass attack to develop fully.

The **B_Main Squeeze** (fig. 10.1) uses the following defaults:

- Lo X frequency = 200 Hz (one octave lower than the default)
- Hi X frequency = 1.5 kHz (one octave lower than the default)
- Attack time = 70 ms (twice as long as the default)

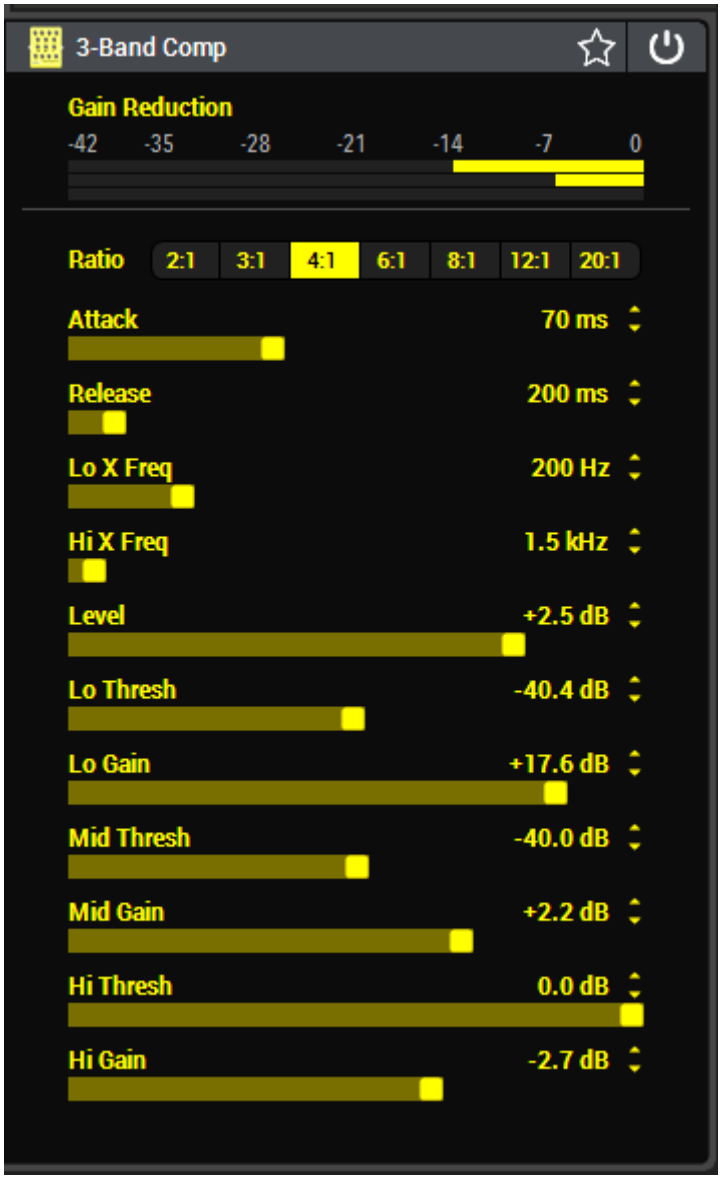


Figure 10.1 With these settings, the 3-Band Comp is voiced for bass instead of guitar.

The Hi band's Threshold parameter of 0.0 dB adds no compression. Instead, it acts like a stage of high-frequency EQ. The Hi Gain parameter controls the level of the highs. More highs make the attack sound more prominent. Fig. 10.1 shows the default 3-Band Comp bass settings, as edited for bass.

However, this is only a default setting. Adjust the Lo, Mid, and High Gain controls for the ideal balance of high, mid, and low frequencies. To compress a band, lower the band's Threshold parameter. This increases sustain but reduces that band's overall output. Compensate by raising the band's Gain parameter.

This effect produces a wide range of bass compression effects, so experiment with the parameters:

- Setting a band's Thresh parameter to 0.0 defeats compression. Then, the band acts like a band in a graphic equalizer, and the Gain control edits the band's level.
- For a bass sound that "snaps," raise the Hi Gain parameter.
- Reduce Mid Gain for a deep, round, "tuned thunder" bass sound that tunnels through a mix.
- For a balanced, pop music-friendly bass sound, reduce the Low Gain.
- Lowering Lo X to 100 Hz gives the low end an almost sub-bass character.

The **B_DI_ProgRockBas** preset (fig. 10.2) is another example of a 3-Band Comp voicing. It gives a tight, bright bass sound, as favored by some prog rock music.

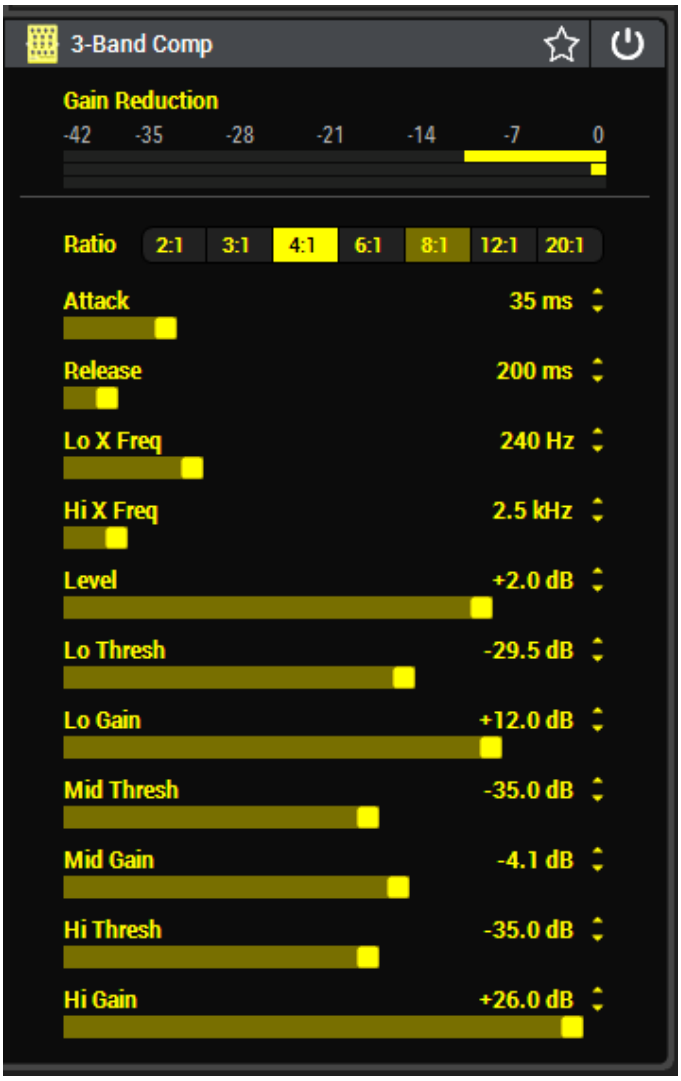


Figure 10.2 There's a lot of compression, but also note the extreme high-frequency boost.

Bi-Amping for Bass

Bi-amping with bass can add effects to the high frequencies while retaining a strong low end. The Input Split block is ideal for bi-amping, because of its frequency crossover mode. In fig. 10.3, the Split (orange line) has a 190 Hz crossover. The Mixer block (yellow line) mixes the low- and high-frequency paths back together.

Unlike the bi-amped guitar effects in this eBook, the Split's Reverse parameter is On. This is a subjective personal preference—I like Path A to be the low-frequency path for bass presets, but the high-frequency path for guitar presets. (I have no idea why—this is not a recommendation!)

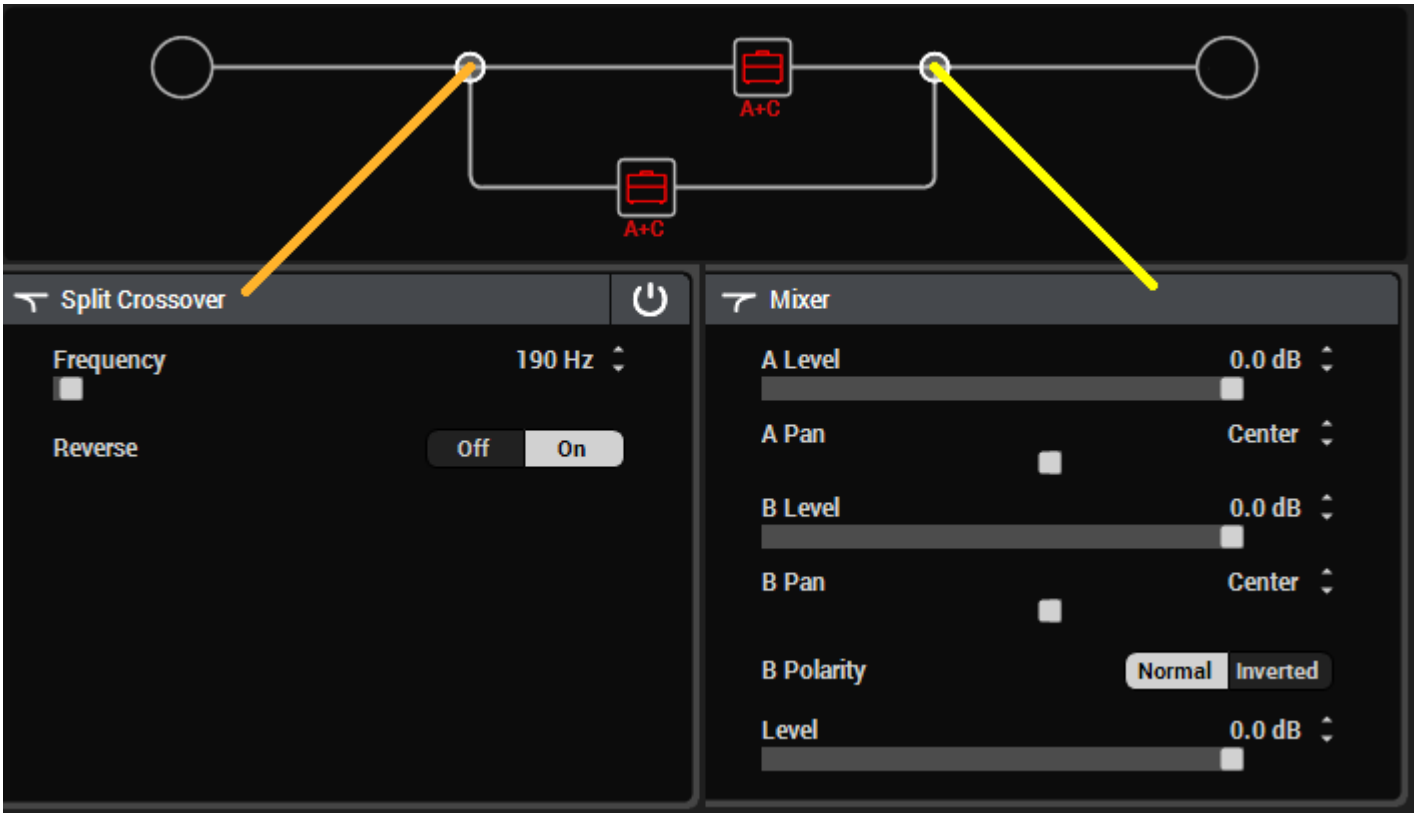


Figure 10.3 Bi-amping is easy with the Split Crossover option. This creates a frequency-selective parallel split.

The preset **B_Bi-Amp** sends the bass to two different Amp+Cabs, so you can optimize the sound for the frequency ranges they cover. Path A is an Ampeg SVT-4 Pro, while Path B is a Woody Blue. Bi-amping is equally flexible whether you want growl on the low frequencies and punch on the highs, or a booming, clean low end with crunched highs.

Bi-amping is a useful technique with effects:

- Compress the lows but not the highs (or vice-versa).
- Add an envelope filter to only the high frequencies for a funk bass sound.
- Spread the highs in stereo, while keeping the low end centered.
- Add tempo-synched effects to the high frequencies, without disrupting the low end.

.

Eight-String Bass

Helix's pitch shifting needs to be tamed for bass. Eight-string bass works best with parallel processing (fig. 10.4). Path A provides a solid bass sound, while Path B creates the "virtual octave higher" strings. The **B_8-String Bass** preset gives a credible, eight-string-bass sound for single-note parts.

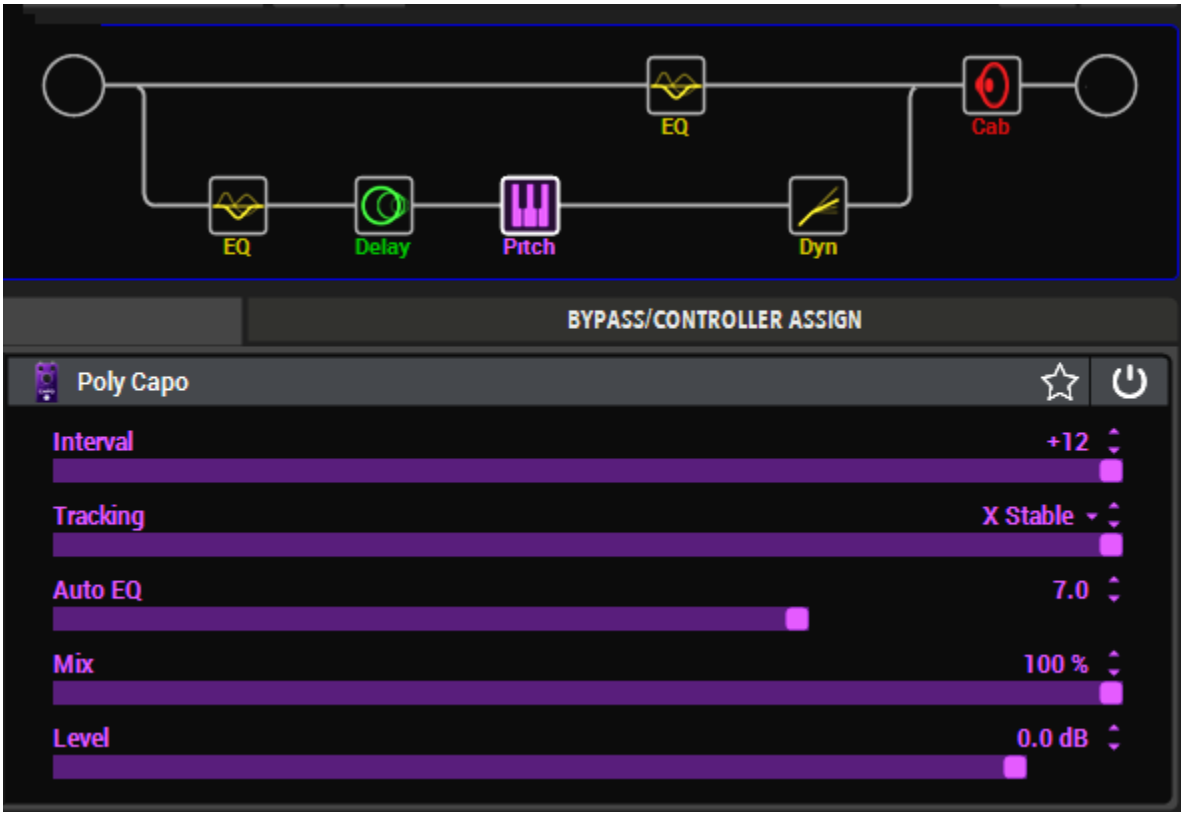


Figure 10.4 The Simple Pitch block provides the transposition needed for an 8-string bass effect.

Path A's simple EQ provides a mild low-frequency boost, coupled with a stronger high-frequency boost to add brightness. Path B for the octave-higher effect has an effects chain:

- Pitch-shifting processors often work best with a clean sound that has few harmonics. The Graphic EQ at the chain's beginning cuts all frequencies to minimum, except for 125 Hz, 250 Hz, and 500 Hz (boosted to +3 dB) and 1 kHz (+7.6 dB boost). It's still best to play cleanly, but these changes keep extraneous frequencies out of the Poly Capo block.
- The Poly Capo block plays *only* the pitch-shifted sound. Its Interval is +12. Mix is at 100% effect.
- With a physical eight-string bass, hitting the higher string of an octave pair occurs slightly after hitting the lower string. To emulate this delay, the Simple Delay has a15 ms delay time. Delays closer to 20 ms may feel more comfortable.
- The LA Studio Comp at the chain's end smooths out the sound.
- The 1x15 Ampeg B-15 cabinet's default settings give a solid bass sound. However, you may prefer an Amp+Cab that gives more of a growl.

Stereo Bass

Traditionally, bass is mono and panned to the stereo field's center. With vinyl, this was necessary to accommodate the physical limitations of vinyl records. Using this approach still makes sense in today's digital world, because bass frequencies are less directional than high frequencies. Most mix engineers pan mono bass in the center, except for EDM-oriented synth bass parts. Spreading these more to the sides allows more room in the center for the kick, snare, and vocals.

However, you *can* have it all: spread the electric bass image in stereo for a wider sound, keep the bass centered, *and* have it collapse properly on mono playback systems. The **B_Stereo Bass** preset (fig. 10.5) gives a "DI" type of sound.

Figure 10.5 Inserting parallel graphic EQs, and using different settings for the left and right channels, provides mild stereo imaging for bass.

Paths A and B have a 10 Band Graphic EQ. The 31.25 Hz, 62.5 Hz, and 125 Hz level settings are the same for both paths, so the lowest frequencies are centered. Higher frequency bands alternate between the left and right channels, which gives stereo imaging. This isn't an exaggerated stereo image, but rather, diffuses the sound somewhat to the sides.

Each path also has an LA Studio Comp with settings similar to the **B_8-String Bass** preset, but with slightly more gain and emphasis.

Virtual Slides

I like to do slides on physical electric bass, but a Helix slide is different. The **B_Bass Slide** preset (fig. 10.6) is a *continuous* slide that ties to your Expression Pedal—not a slide that moves in semitone jumps. This gives a synth bass vibe.

The Low and High Cut EQ at the end trims highs so that when sliding up, the sound isn't overly bright. However, if you bypass the EQ, you can leave this effect enabled, because the effect doesn't affect the sound unless you're shifting the pitch. The Position parameter defaults to expression pedal EXP 1.

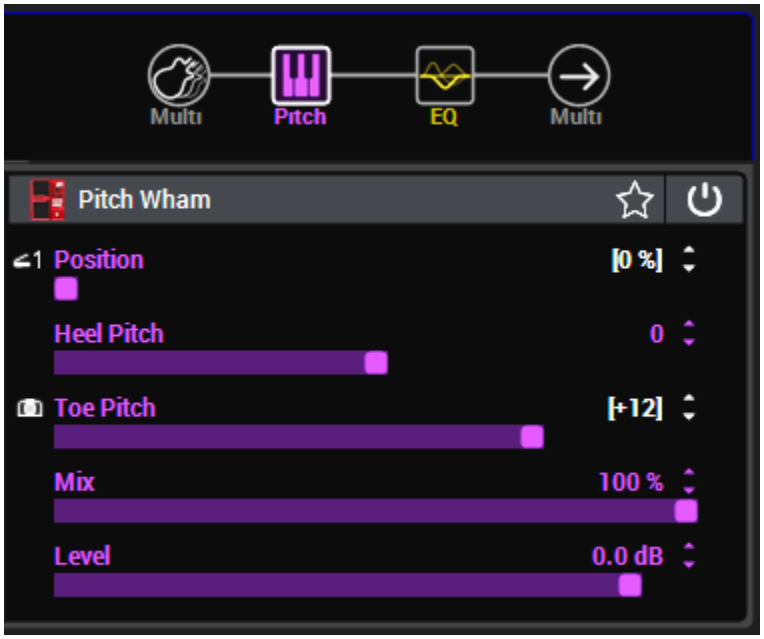


Figure 10.6 Preset for virtual bass slides.

At least for me, being able to slide either up *or* down is a non-starter. It's difficult to return the pedal to center (i.e., go back to no pitch change) when the pedal covers a wide range of frequencies. One solution is to create two presets, one that slides up with maximum Toe Pitch at +12 semitones, and one that slides down with maximum Toe Pitch at -12 semitones. However, the **B_Bass Slide** preset uses Snapshots instead:

- Snapshot 1 slides up one octave (+12 semitones)
- Snapshot 2 slides down one octave (-12 semitones)
- Snapshot 3 slides up a fifth (+7 semitones)

The preset has 3 Snapshots for HX Stomp compatibility. With a 4-Path Helix, the 8 Snapshots could all shift pitch to different intervals.

Tip: Helix can do two-octave slides, but I don't find that as useful as one-octave slides. Two-octave slides make precise pitch changes with the pedal difficult.

Funk Filter

Envelope-controlled bandpass filters work well with bass, if they're in a parallel path so that the bass maintains the low end. However, you'll need to edit the Autofilter's Filter Q and Sensitivity to accommodate your playing. Someone snapping strings hard will need different settings compared to someone who plays with their fingers or a pick. The **B_Funk Filter** preset (fig. 10.7) is a good place to start, but will require tweaking.

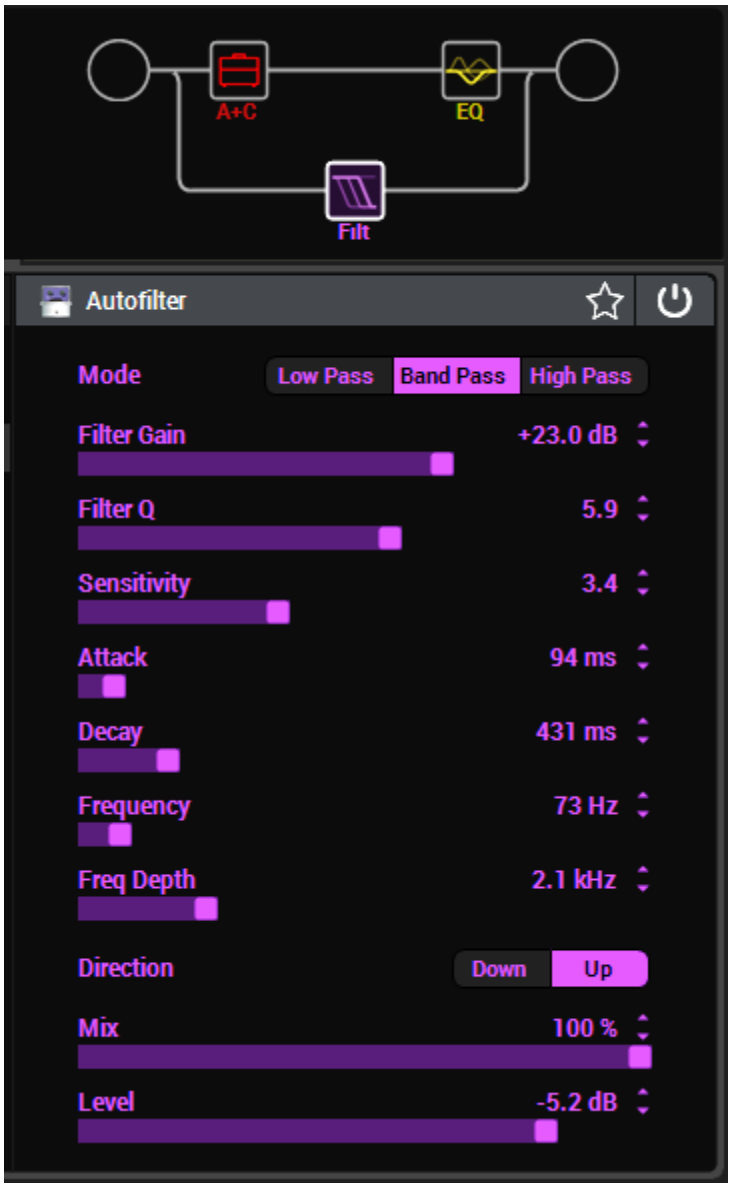


Figure 10.7 The Autofilter goes in parallel with your Amp+Cab of choice.

The Ampeg B-15 Amp+Cab gives the right amount of grit to my ears, but you may want a cleaner or more aggressive sound. The 10 Band Graphic EQ in Path A lowers the 2 kHz and higher-frequency sliders. This carves out part of the frequency spectrum to make room for the Autofilter's mid and upper mid frequencies in Path B.

DI (Direct Injection) Sounds

These presets are designed for solid, basic sounds, with little obvious processing.

B_DI_CmpresHighs uses the Input Split as a crossover. Frequencies below 300 Hz go to a Simple EQ that brings up the low frequencies slightly. Frequencies above 300 Hz go to an LA Studio Comp, set up as a limiter to squeeze the high frequencies. The limiting function can also tame out-of-control slap bass peaks.

B_DI_Pnch+MidDrp has two effects. The LA Studio Comp's Type parameter is set to Limit, not Compress, to keep peaks under control. That's followed by a 10 Band Graphic (fig. 10.8) to reduce the mids, boost the low end, and boost around 2 kHz.

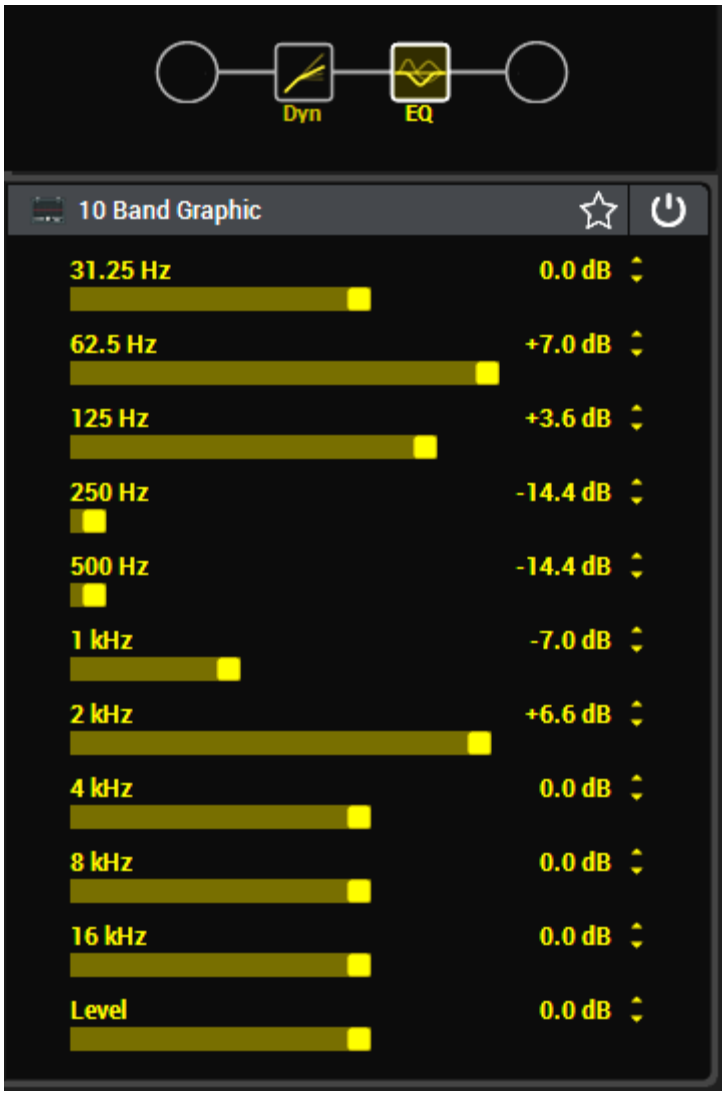


Figure 10.8 The EQ settings drop the mids. This makes room for midrange instruments, while keeping the low and highs.

B_DI_Pnch+MidBig is like **B_DI_Pnch+MidDrp**, but takes an opposite approach that shows how changing EQ can make a big difference in the sound. The boost at 1 kHz helps bass cut through a mix.

Hollow Bass

Some bass players, particularly jazz fans, use chorus units to make the bass “swim” a bit. Preset **B_Hollow Bass** (fig. 10.9) is for those who want the bass to “float” instead of swim. The tone is similar, but the effect is more static.

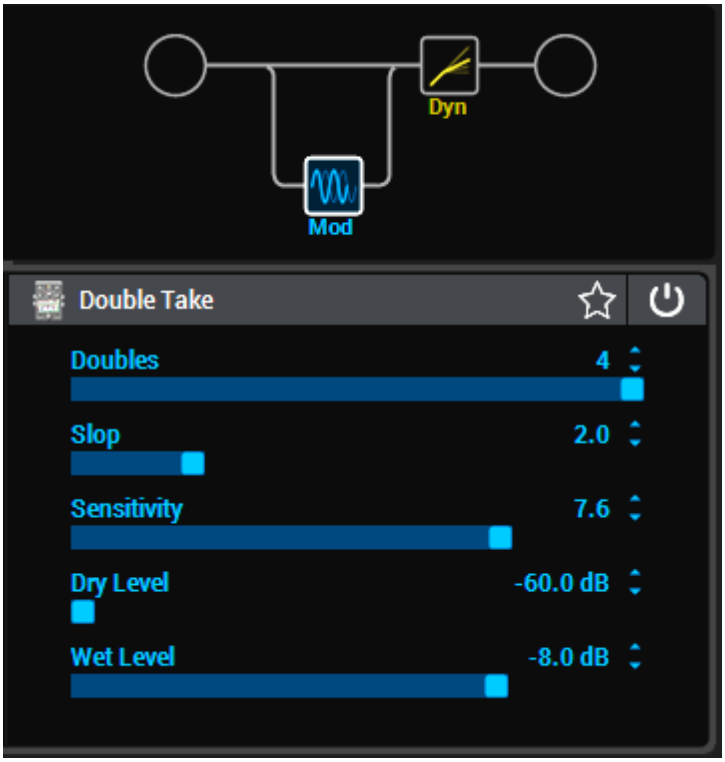


Figure 10.9 This gives bass somewhat of a more ethereal vibe.

The reason the tone is more static than a chorus is because the Double Take block provides the effect. The Double Take doesn’t have an LFO, so it doesn’t create the continuous changes of a traditional chorus.

Wet Level is the most important parameter. The higher the value, the “hollower” the sound.

Rhythmic Patterns for Bass

Asheville Pattern adds rhythmic, filtered effects to bass. However, programming its parameters can be a little confusing—so let's unpack the **B_Rhythm Filter** preset. This effect is most obvious with higher frequencies. To learn what it does, it's helpful to create a preset with the Asheville Pattern in parallel with the main bass sound (fig. 10.10).

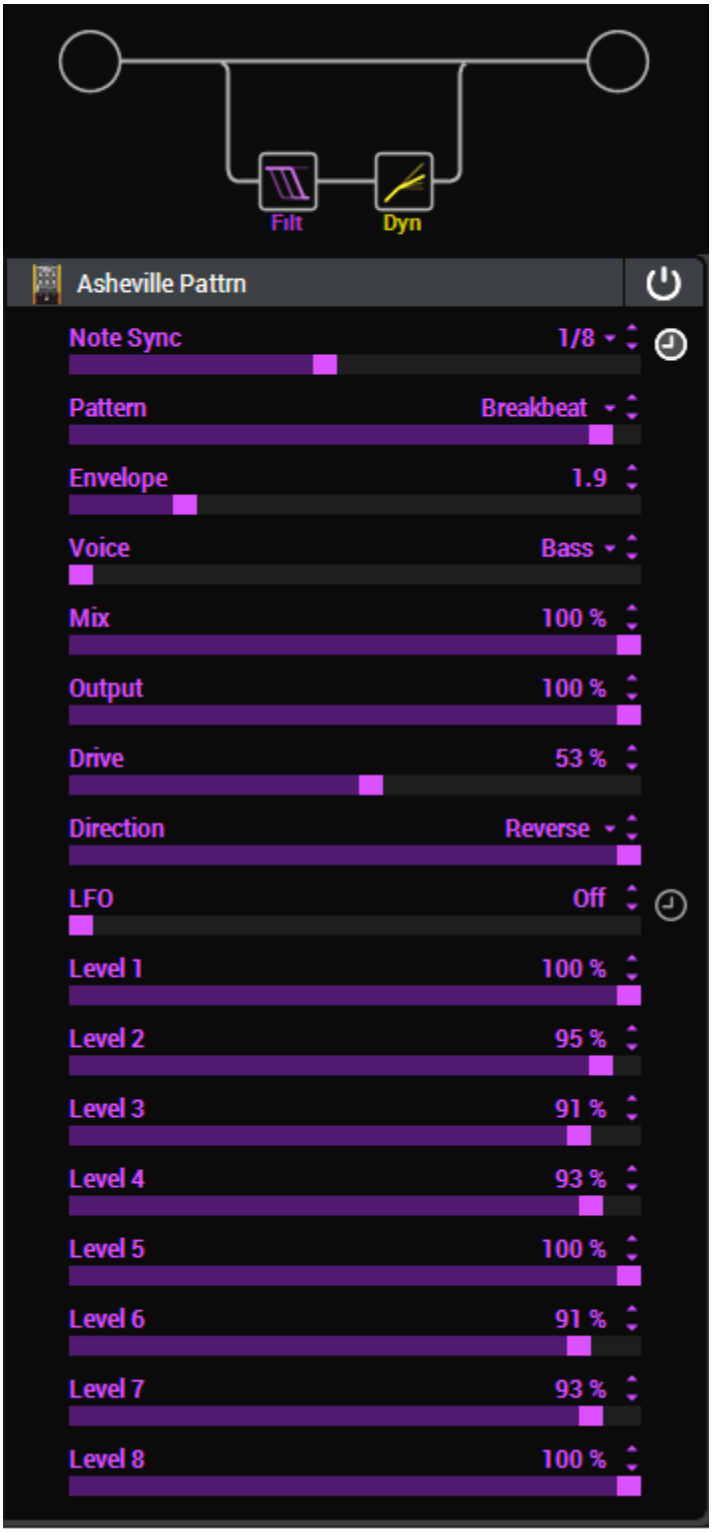


Figure 10.10 Basic Asheville Pattern preset for bass.

Turn up the effect's Drive parameter so that the bass has plenty of highs. Or, insert a distortion like the Heir Apparent after the Input Split, and prior to the Asheville Pattern.

Tip: You can't use the Ashville Pattern's Mix control to provide a true dry/wet parallel effect, because the dry Mix setting will include any added Drive in the Ashville Pattern effect. You may not always want the extra grit.

To optimize the filter frequencies for bass, set the Voice parameter to Bass. However, when using Ashville Pattern in parallel with a dry bass sound, the Mids setting can be useful too. I also recommend following the Ashville Pattern with an LA Studio Comp to smooth out volume variations. The default settings work well, but also try lower Peak Reduction values.

The Pattern effect has 8 filters. The effect either alters the selected filter levels with an LFO or uses a sequencer that steps through the filters. Filter 1 is a lowpass filter, which preserves the bass's characteristic low-frequency energy. Filters 2-7 are bandpass filters, each tuned to a successively higher frequency.

LFO Mode

This is the simpler of the two modes. To choose LFO mode, set the Pattern parameter to Off. The Envelope, Direction, and Rate (called Note Sync if its sync option is enabled) parameters have no effect in this mode. The LFO parameter is a free-running LFO that sets the effect speed in LFO mode. When synched to tempo, the LFO label changes to Note Sync.

Start with Level 1 at 100%, and the other Levels at 0%. Play a sustained bass note. (With the Helix Native plug-in, it's convenient to record a sustained note, and loop it while you explore this effect's settings.) Now bring various filters up to 100%, and then back down to 0% again.

Each filter has its own influence on the sound. For mild phase shifter effects, bring up some filters, but not others—for example, set Levels 3, 5, and 7 to 100%, while leaving 2, 4, and 6 at zero. Remember, you'll hear a prominent, obvious effect *only* if the effect can process high frequencies (like what occurs with Drive turned up, or with a distortion effect inserted before the Asheville Pattern).

As you vary the filter Level settings, also experiment with the LFO speed.

Pattern Mode

Patterns are the key to driving, rhythmic bass sounds, especially when synched to drums.

Start by turning the LFO parameter to Off. Choose Stair for the Pattern and set the Rate for around 5.0 to 6.0 (with Note Sync, try 1/8th notes). Choose around 2.5 for Envelope. Turn up all the filter level controls, and you'll hear a higher filter frequency on each successive Pattern step, until it jumps back to the beginning.

Now, vary the Envelope parameter. Lower values give a more percussive envelope decay. Increasing the value shortens the envelope attack up to a point, after which the sound becomes more percussive again. This is a fun parameter to assign to an Expression Pedal.

Next, audition the various Patterns. I prefer Rhythmicon, Breakbeat, and Big Beat, because they add percussive accents that help drive songs. Use the Direction parameter to choose the pattern direction. For example, with the Reverse setting, the Stair pattern goes from the high filter to the low filter, instead of the usual low-filter-to-high-filter pattern.

One of my favorite “EDM bass” sounds is to set the Envelope for a short decay, and turn up only some of the filter levels. This gives an interesting, syncopated sound. If Asheville Pattern doesn’t start on the beat, reset it by hitting the Tap Tempo footswitch on a measure’s downbeat. Once synched, whatever pattern you dial up stays in time with the tempo.

Bass Vibrato

Why should only *synth* bass players have all the fun? It's difficult to create guitar-type vibrato, because bass strings are so thick. However, we can add electronic vibrato with the **B_Bass Vibrato** preset (fig. 10.11).

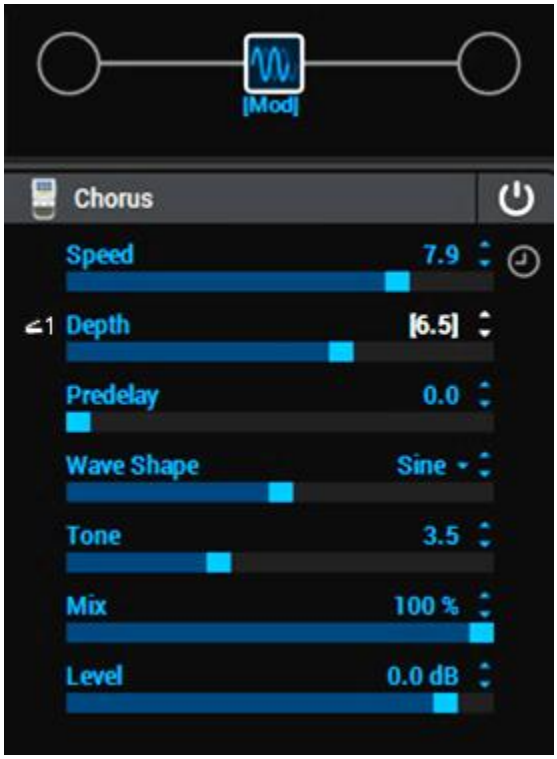


Figure 10.11 The Helix chorus can create an excellent vibrato effect for bass.

Helix's Legacy effects include a pitch vibrato, but I prefer the standard Chorus for this application. The Chorus is the only block that's needed. Set it for 100% processed sound. Control Depth with Expression Pedal EXP 1.

Like a keyboard synthesizer's modulation wheel, the pedal provides expressiveness because you can bring the vibrato in or out as appropriate. A maximum pedal depth of 6.5 seems about right. With bass, I prefer lower Tone parameter settings for the Chorus—but choose whatever setting puts a smile on your face.

Bass/Drum Grooves

The **B_BassDrumGroove** preset (fig. 10.12) provides rhythmic pulsing that complements drum grooves. The Speed parameter is the base tremolo rate. With tempo sync, the Speed parameter's name changes to Note Sync.

Like the Asheville Pattern block, when synched to tempo the Bleat Chop Tremolo doesn't always start on the beat. A quick press on the Tap Tempo button, on the downbeat, will sync the Bleat Chop Trem. The two LA Studio Compressor blocks smooth the sound. They're set close to their defaults.

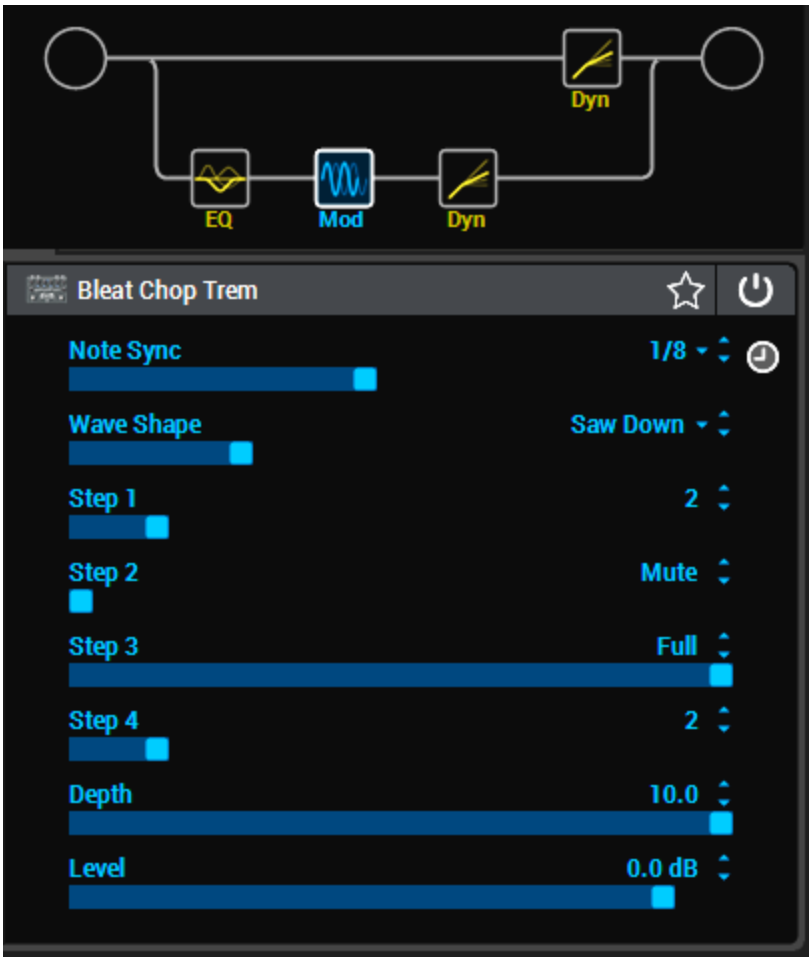


Figure 10.12 The Bleat Chop Tremolo has more rhythmic complexity than standard tremolos.

Four successive steps create the groove. Each step has a tremolo speed multiplier. Additional steps for mute and maximum level are on the left and right ends of the Step sliders respectively.

Setting all the Step parameters to 1 creates a standard tremolo, running at the base tremolo speed. I like using Sine or Saw Down for the waveform.

Setting all the step parameters to 2 doubles the tremolo speed. The most interesting tremolo patterns happen when all the steps *aren't* the same. For bass, I rarely use an option other than 1, 2, mute, or max. The super-fast tremolo speeds can sound cool with guitar, but they interfere with the bass's solid foundation.

When you start editing the parameters, change one of the steps to Mute, then change another Step to 2. Try other combinations to end up with various hypnotic, rhythmic effects. The Saw Down Wave Shape's quick decay can even give an EDM-friendly sound.

Big Growl

The **B_Big Growl** preset (fig. 10.13) is more like a template than a preset, because its parallel amps can provide a wide variety of tones. Options include adding distortion to taste with the amp drive controls, changing out different amps, and auditioning different cabs. You can also separate the Amp and Cab blocks. This allows using a Dual Cab block after the Amp.

Path A's 8x10 SVT AV Cab uses the 4038 ribbon mic. It's known for giving a full, booming bass tone. Path B is more about the distortion aspect. So, varying their balance in the Mixer changes the growl's character.

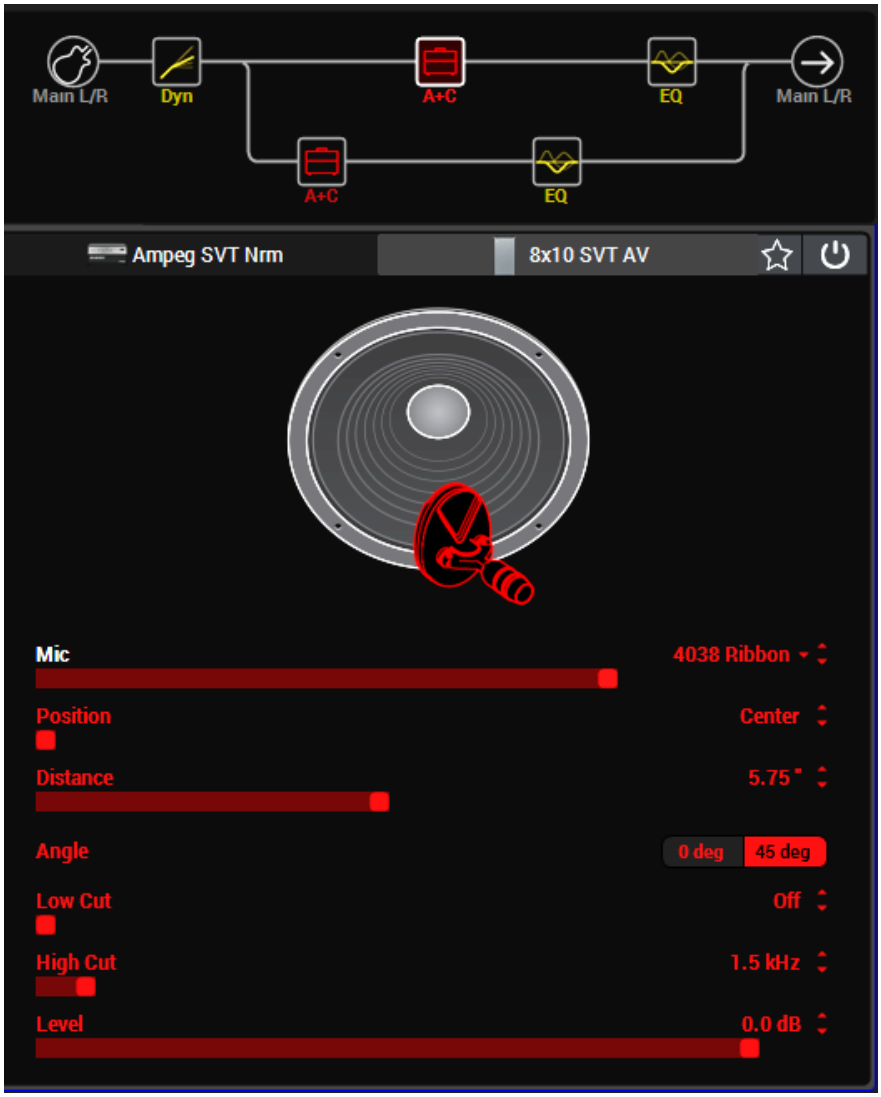


Figure 10.13 Paralleled amps and cabs give an immense amount of tonal flexibility.

The 10 Band Graphic EQs at the path outputs are optional. They provide a stereo spread, as described in the application Stereo Bass. Even with the stereo spread, the bass's lower frequencies remain centered, and the stereo image collapses perfectly to mono.

Sound Design with Bass

Bass can also do cool sound effects—the **B_Explosion** preset creates sounds like explosions, thunder, and rocket exhausts. It's also useful for adding emphasis when a song hits The Big Chorus, as well as breaking leases or scaring small animals.

- Aside from the Noise Gate Threshold parameter, you need to change the defaults only for the Dual Pitch and Low and High Cut blocks (fig. 10.14).
- To free up a block, use the Input Gate instead of a Noise Gate.
- The reverb uses Ganymede's default setting.
- For distortion, I like Teemah! and Deranged Master, but all of them are candidates. Turn up any distortion drive controls for even ruder sounds.

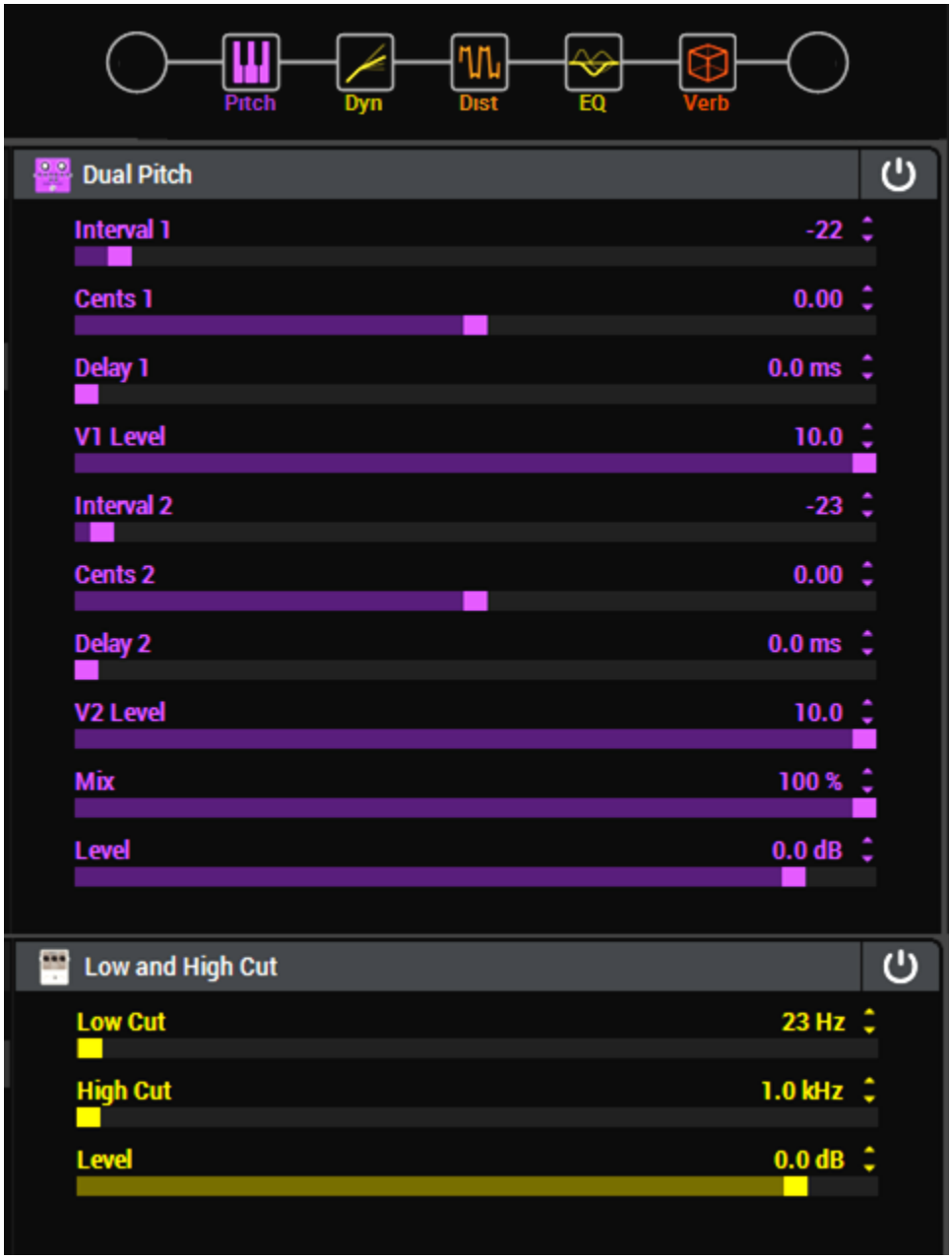


Figure 10.14 Frighten small animals, or score movies that need an intergalactic cosmic explosion, with this preset.

Mr. Saturation

The preset **B_MrSaturation** delivers mild distortion. It's a simple preset—the distortion block in parallel Path B feeds into an Ampeg B-15 Cab.

However, there is a secret ingredient. The Vital Boost distortion effect has only a Boost parameter, but it's controlled by EXP 1. This makes it easy to add expressive saturation with the pedal. The more forward the pedal, the more saturation. Simultaneously, this reduces the cab's output level (fig. 10.15). The goal is to avoid excessive level swings when going from less to more saturated sounds.

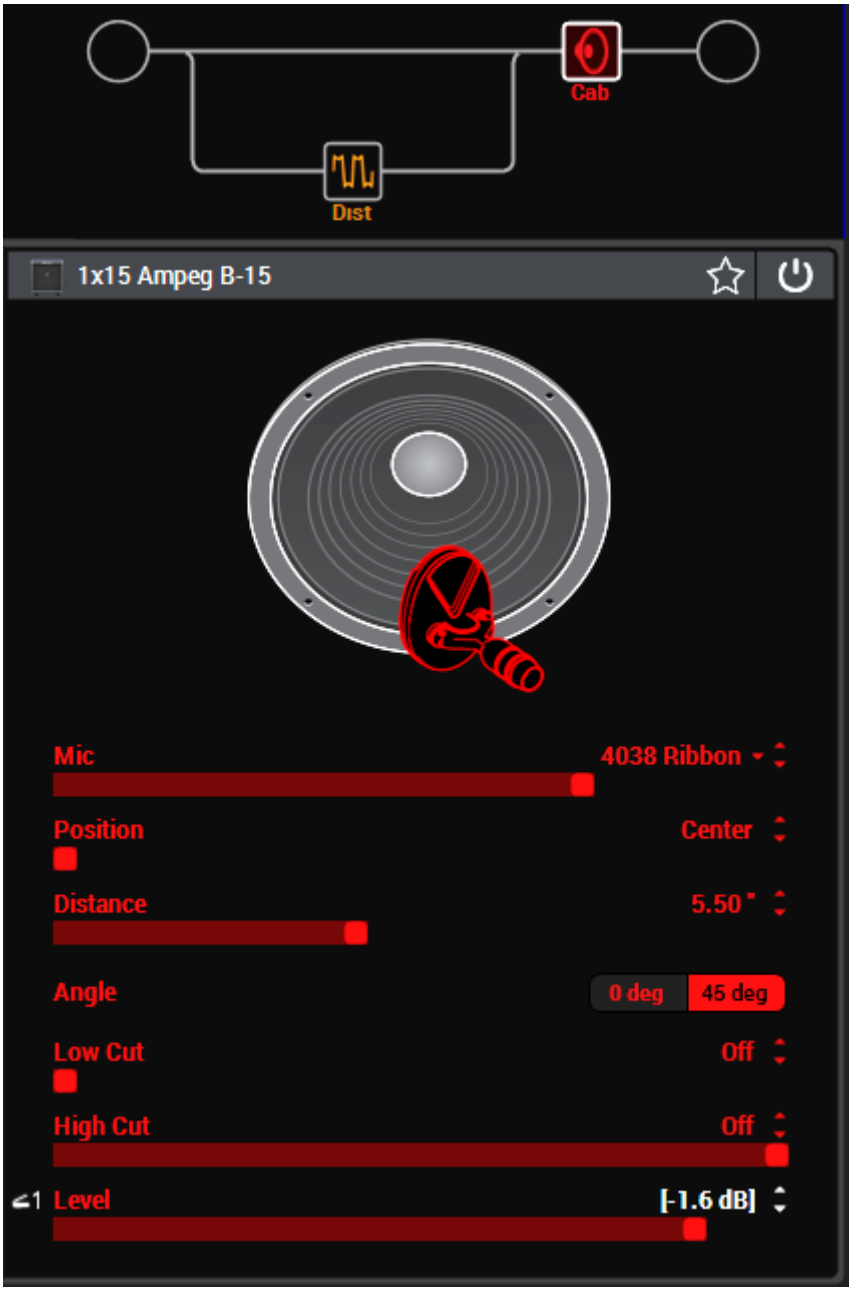


Figure 10.15 Tying the Cab level to the Boost amount avoids excessive level swings.

Rude Bass

To go past mild saturation into nastier territory, try **B_RudeBass**. A Deluxe Comp increases the sustain before splitting into two parallel paths. These use 3-Band Comps to create a Combi-Band frequency split. Path A isolates the midrange, while Path B isolates the low and high frequencies (fig. 10.16). Each path has an Alpaca Rouge distortion.

Path A feeds an Ampeg B-15NF amp and 1x15 Ampeg B-15 Cab. Path B uses a Del Sol 300 Amp and 2x15 Brute Cab. Path B provides the majority of the final sound, because it carries the low and high frequencies. The Simple EQ at the output provides a tone stack.

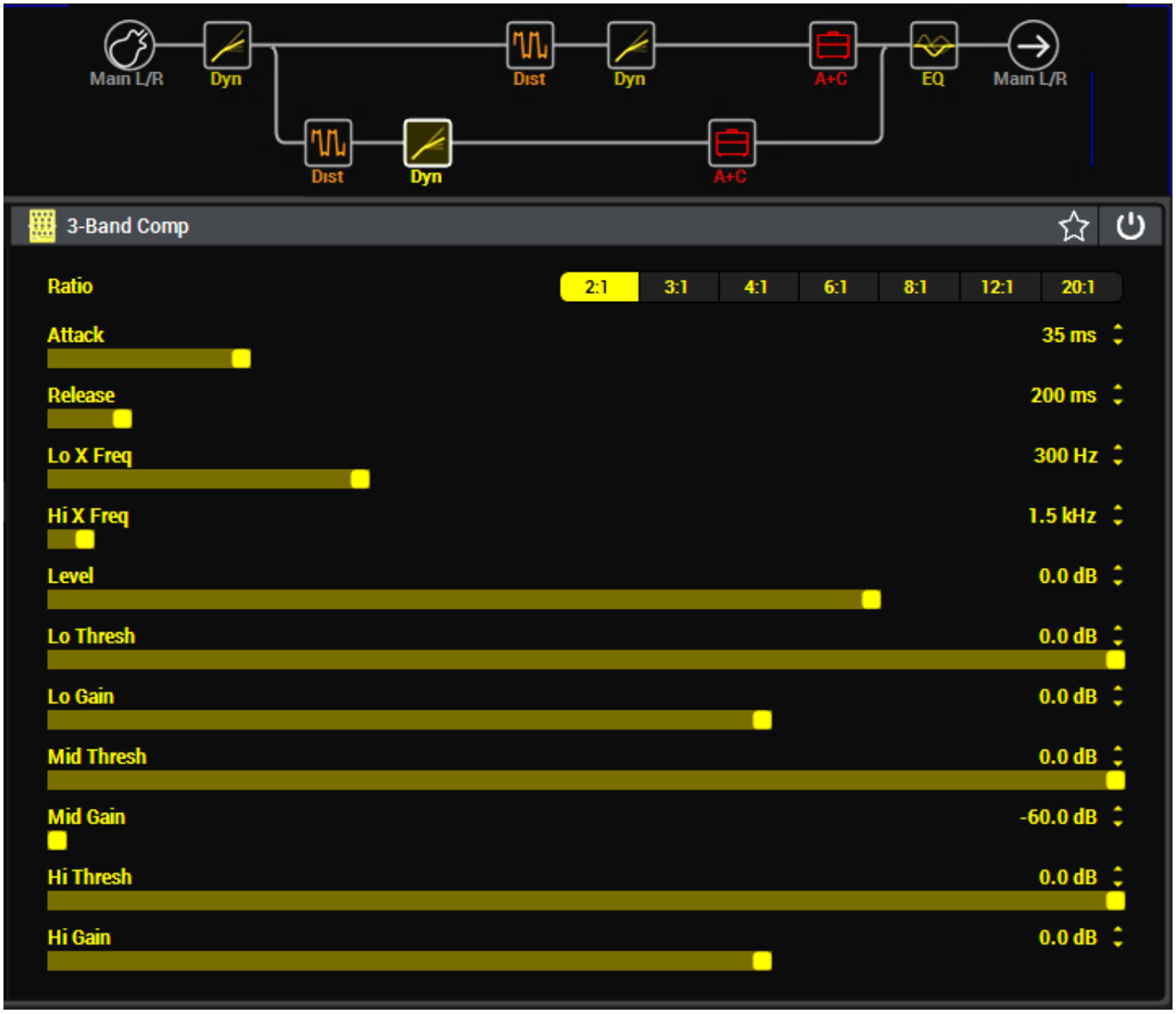


Figure 10.16 Path B's 3-Band comp (shown above) isolates the low and high frequencies by turning the Mid Gain band all the way down. Path A's 3-Band Comp isolates the midrange by turning the Mid Gain up all the way, while turning down the Lo and Hi Gains.

Wide Bass

In the studio, bass is almost always centered, and *has* to be centered if it will be mixed for a vinyl release. The preset **B_Wide Bass** (fig. 10.17) provides a stereo spread, but does so without compromising the main bass sound in the center.

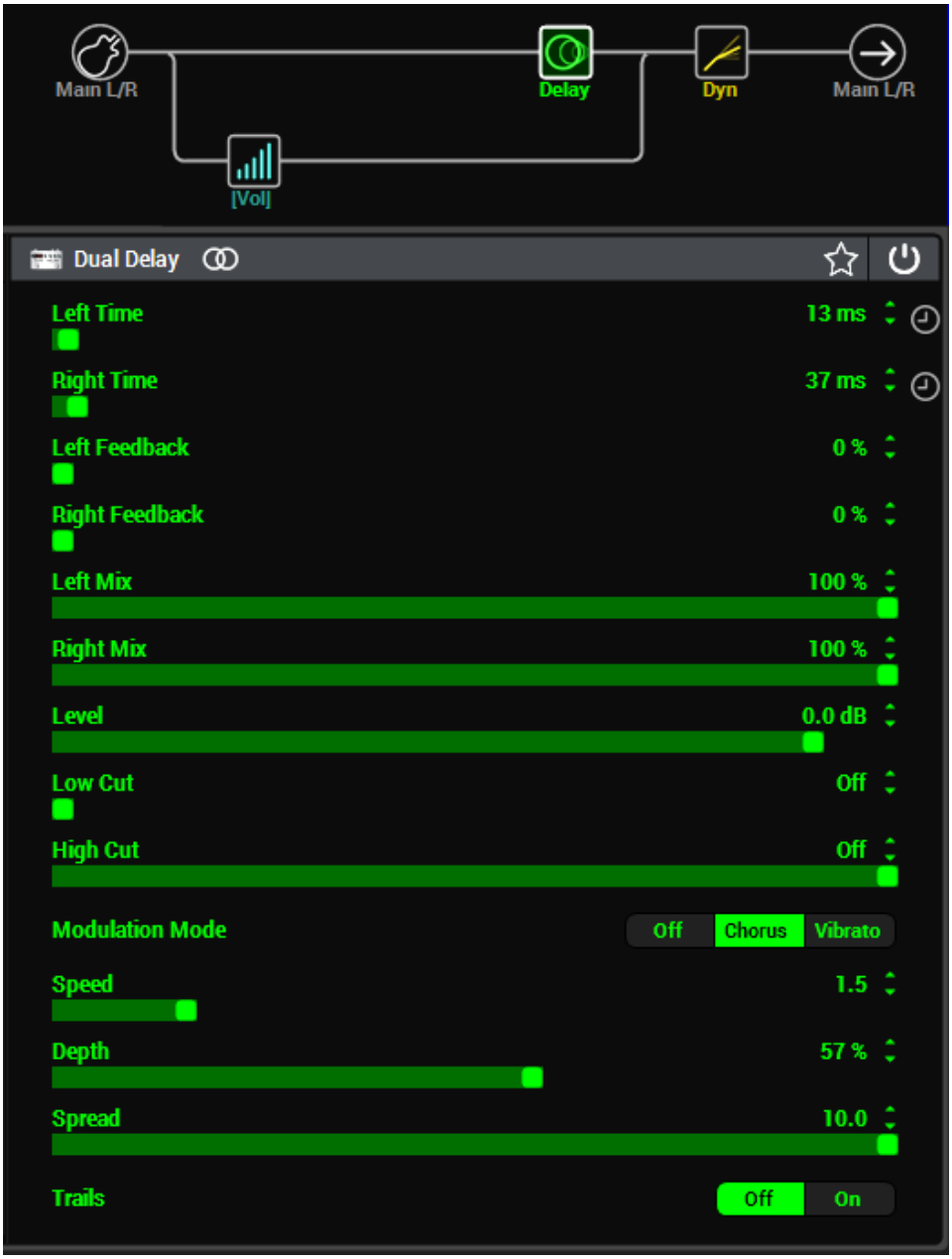


Figure 10.17 The Split Crossover and Dual Delay make this preset possible.

The Input Split Crossover at 370 Hz splits the high and low frequencies into Paths A and B, respectively. The Vol block in Path B has a gain of 0.0 dB, because all it does is create a parallel path for the low frequencies. The highs proceed through a Dual Delay, which adds slightly different delays for the left and right channels. This spreads out the highs into stereo, without interfering with the low frequencies.

Enabling Chorus in the Dual Delay is optional. With Depth turned up, chorusing adds some animation to the high frequencies. The output Mixer lets you choose a blend of the high and low frequencies.

Bass Room

B_Bass Room gives the vibe of playing in a rehearsal room, small club, or similar space. Like most other presets in this eBook, it obeys the rule of "keep the bass centered."

Referring to fig. 10.18, the main bass sound splits into Path A, which drives an LA Studio Comp dynamics processor for more consistent dynamics. The other split goes to Path B, which uses the Dynamic Ambience reverb to provide the sound of room reflections. To avoid a muddy reverb sound, the 10 Band Graphic before the reverb block reduces the low frequencies. So, only the highs end up being reverberated and spread in stereo.

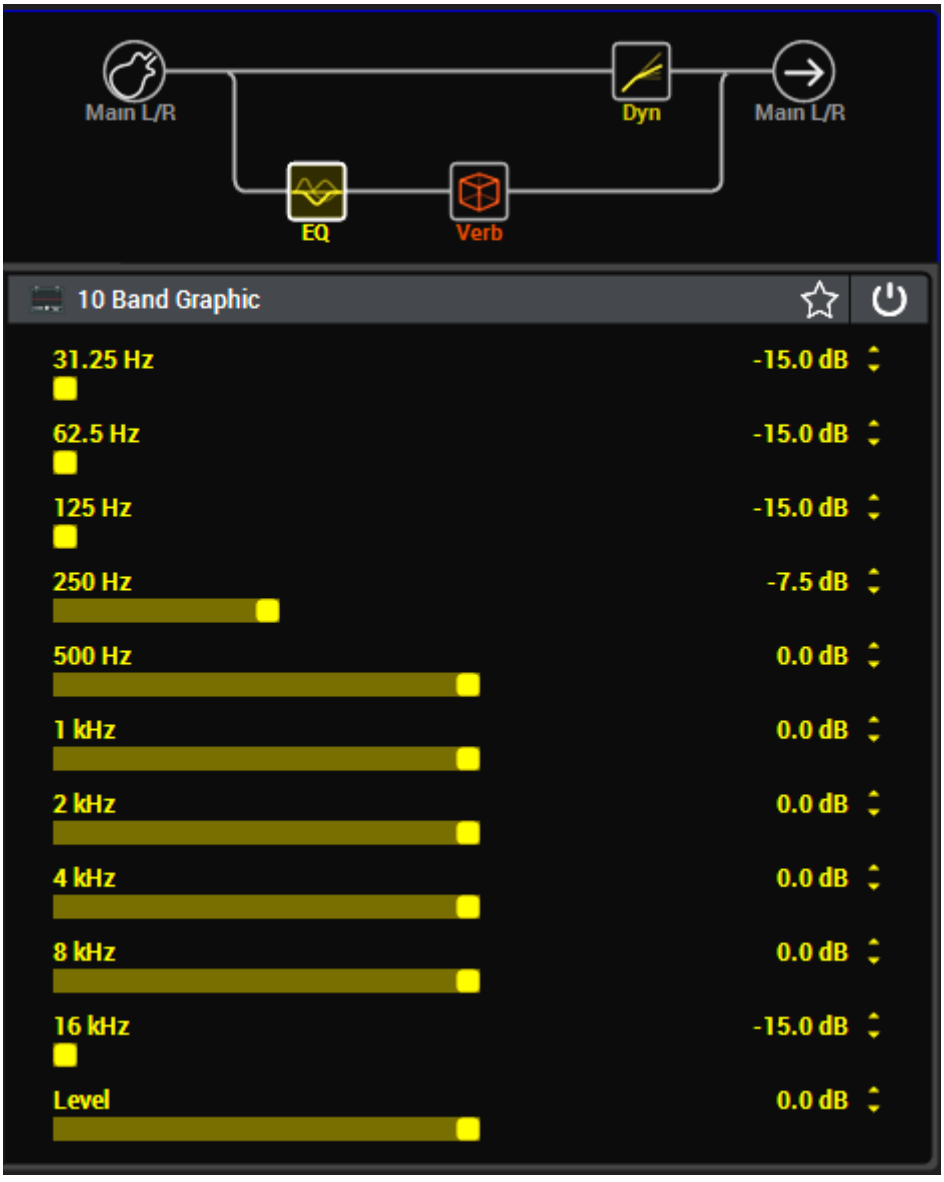


Figure 10.18 Restricting the frequencies going into the Dynamic Ambience processor keeps the bass reverb from getting muddy.

Slapback Bass

Slapback echo isn't only for vocalists and guitar players. For sparse bass parts, a little slapback echo can impart a vintage rockabilly feel.

The preset **B_SlapbackBass** (fig. 10.19) is basic. The Simple EQ gives a moderate low-end boost. The Simple Delay provides the slapback effect. The Delay time is the slapback echo time used at Sun Studio for recordings by Elvis Presley, Carl Perkins, and others. If you turn up the mix much more than 20-30%, you'll probably want to reduce the Time parameter somewhat so that the echo doesn't "step on" the bass.

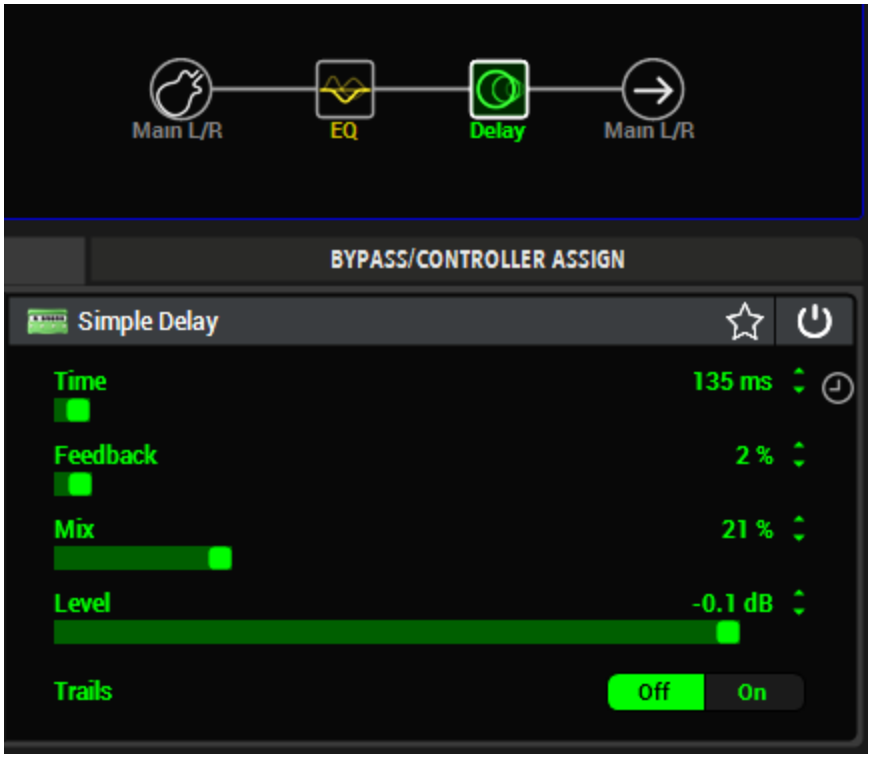


Figure 10.19 Slapback echo isn't used much on bass, so adding it gives a novel sound and a different vibe.

Wah Bass

B_Wah Bass defers to the core bass sound by subtly layering the wah sound on top (fig. 10.20). The effect is intended to sound more like a variable tone control than a stereotyped wah pedal.

Several of the Wah models let you set high and low frequency limits, so you can tune them for bass. I chose the Vetta because I like the way its bandwidth and resonance characteristics work with bass.

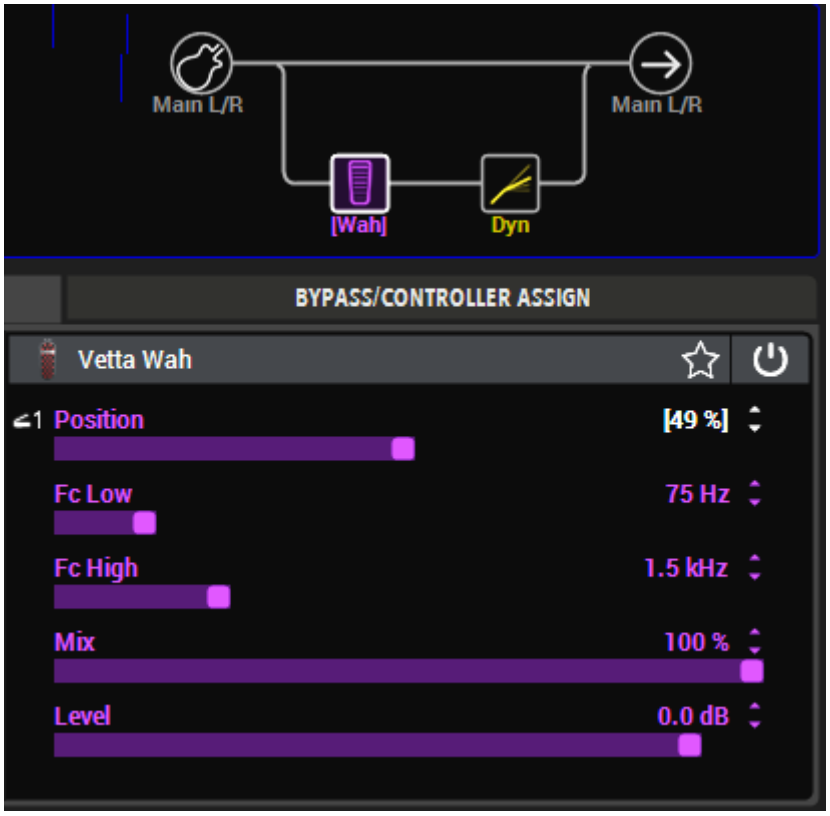


Figure 10.20 Inserting the Wah in parallel keeps the low end intact.

The Wah position is tied to Expression Pedal EXP 1. Given that the Wah has a Mix parameter for dry/wet sound, the parallel path may seem redundant. However, separating the Wah sound allows following it with the LA Studio Comp dynamics processor, with Type set to Limit instead of Compress. This tames any peaks caused by the Wah's frequency boosts. Instead of varying the dry/wet balance in the Wah, alter it in the output Mixer.

Focusing-Type EQ

This deceptively flexible preset gives a solid, customizable bass tone. **B_Focusing EQ** takes advantage of the 3-Band Comp's ability to focus on a wider range of frequencies compared to typical peaks and notches.

Fig. 10.21 shows how to set up the 3-Band Comps. The input feeds a parallel split. Each split has a 3-Band Comp block. Path A is the low-frequency path. Path B is the high-frequency path. To use the 3-Band Comp for EQ instead of compression, set all Thresh parameters at 0.0. If all the Threshold parameters are at 0.0 dB, the Attack, Release, and Ratio parameters don't matter.

Path B's most important parameters are colored light blue. Lo X Freq sets the high band's lowest frequency, while Hi X Freq sets the high band's highest frequency. Mid Gain adjust the high band's level. Path A's most important parameters are colored white. Lo X Freq sets the low band's highest frequency (there's no need to set a lower frequency limit). Lo Gain adjusts the low band's level.

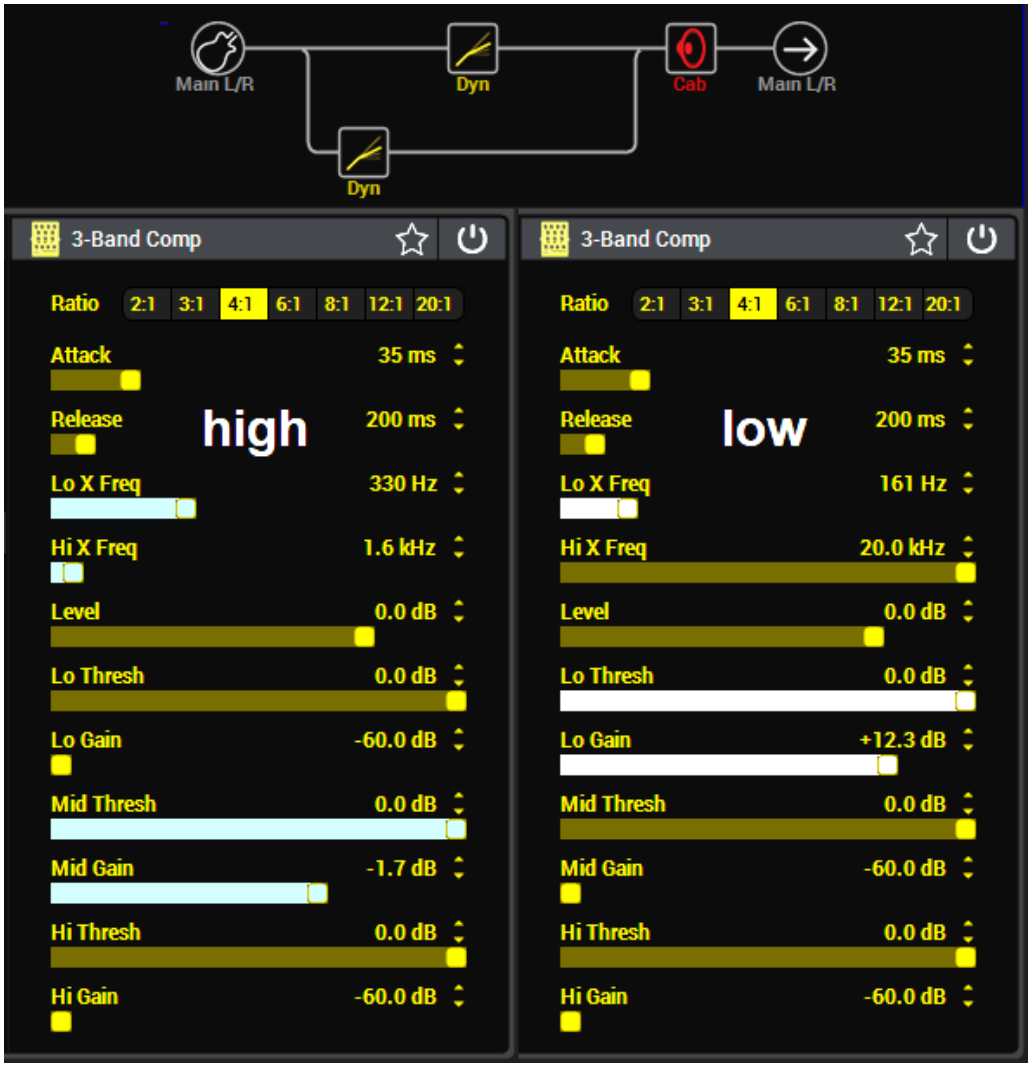


Figure 10.21 The Lo and Hi X Freq parameters choose the ranges that will get the EQ's focus. The Gain parameters choose the balance of the two ranges.

The 2x15 Brute Cab is a good choice because it covers a wide frequency range. For many applications, the 421 Dynamic mic works well.

Fifth Bass

B_FifthBass layers a fifth above the bass, in parallel with the main sound. This preset shown in fig. 10.22 includes an Amp and Cab, but the preset doesn't need them unless you want an amp sound. (I almost always use this preset without the Amp and Cab. The clean transposition is a great effect.)

The Low and High Cut EQ improves tracking slightly by restricting the frequency range going into the Poly Capo. Most of the time, the Poly Capo goes at the beginning of a chain of effects. So, placing the Poly Capo between the Cali 400 Ch1 Amp at the chain's beginning and the 8x10 SVT AV Cab at the end is somewhat unusual. However, this seems to give the best results.

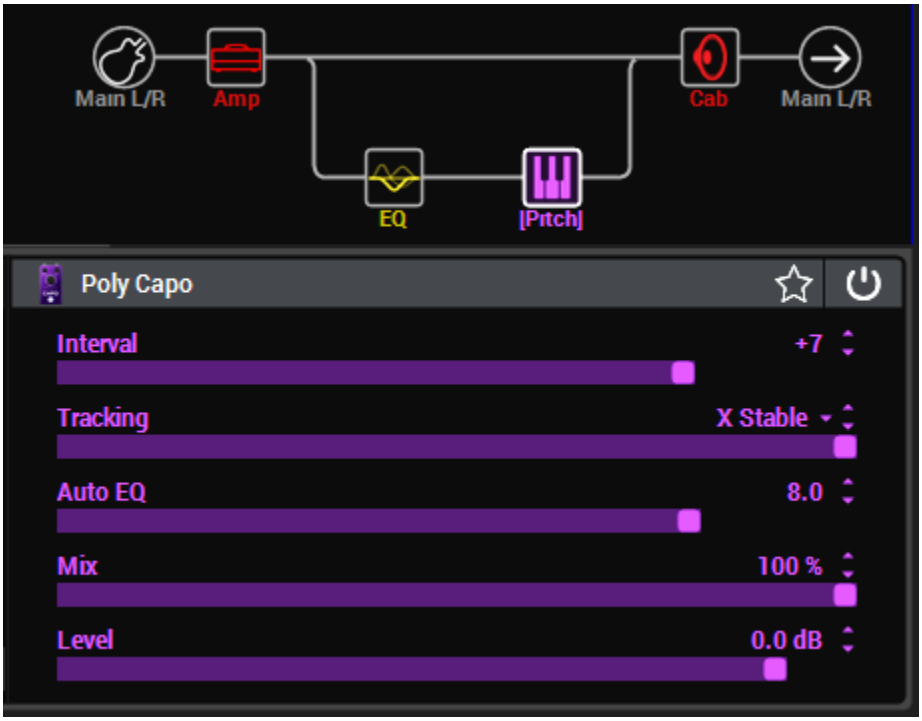


Figure 10.22 The Poly Capo is in parallel with Path A.

The Poly Capo has a dry/wet mix control, so the parallel placement may seem redundant. However, this preset lends itself well to controlling Path B at the mixer with Expression Pedal EXP 1. Then, the dry level stays constant, and the pedal augments the main sound with the fifth above.

Fifth LCR Bass

This preset is all about ***Drama!*** It's related to the previous preset, but **B_FifthLCRBass** splits into standard bass tuning, and a bass sound a fifth above. Both paths go through Amps and Cabs (fig. 10.23).

The reference to LCR mixing is because the standard bass is panned hard left, and the fifth above is panned hard right. You probably don't want this preset active all the time. But in the studio, switching over to this preset while a lead part is centered produces a novel, dramatic effect.

Like **B_FifthBass**, the Poly Capo inserts between the Amp and Cab. The Low and High Cut EQ before the Poly Capo cleans up the attack slightly.

Note that the Amp and Cab you choose for Path B is crucial. Because the transposed sound is brighter, you'll probably want a Cab and mic that reduce the highs somewhat. This preset uses an 8x10 SVT AV Cab, with a 4038 ribbon mic to mellow out the tone.

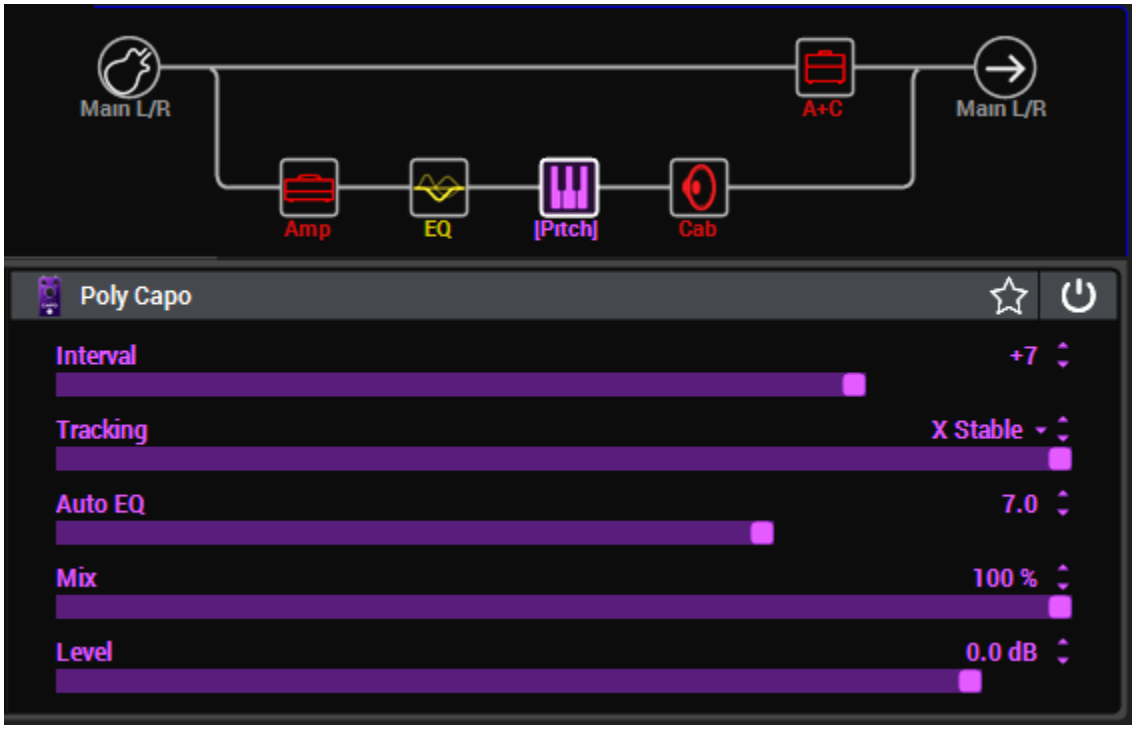


Figure 10.23 This variation on the previous preset is most effective in the studio.

Dancing Highs in Stereo

Let's move on to some 4-Path Helix presets. Most of the time, I'm happy with straightforward bass processing, like some EQ and dynamics. However, with simpler musical arrangements, incorporating some ear candy in existing parts can add sufficient variations that you may not feel the need to add more parts.

The preset **B_DancingHighs** (fig. 10.24) splits Path 2's high frequencies into dual Harmonic Tremolos, set to slightly different speeds. The result is a stereo spread that moves in the stereo field, while sitting "above" the centered bass part. The effect is subtle but adds motion to the sound.

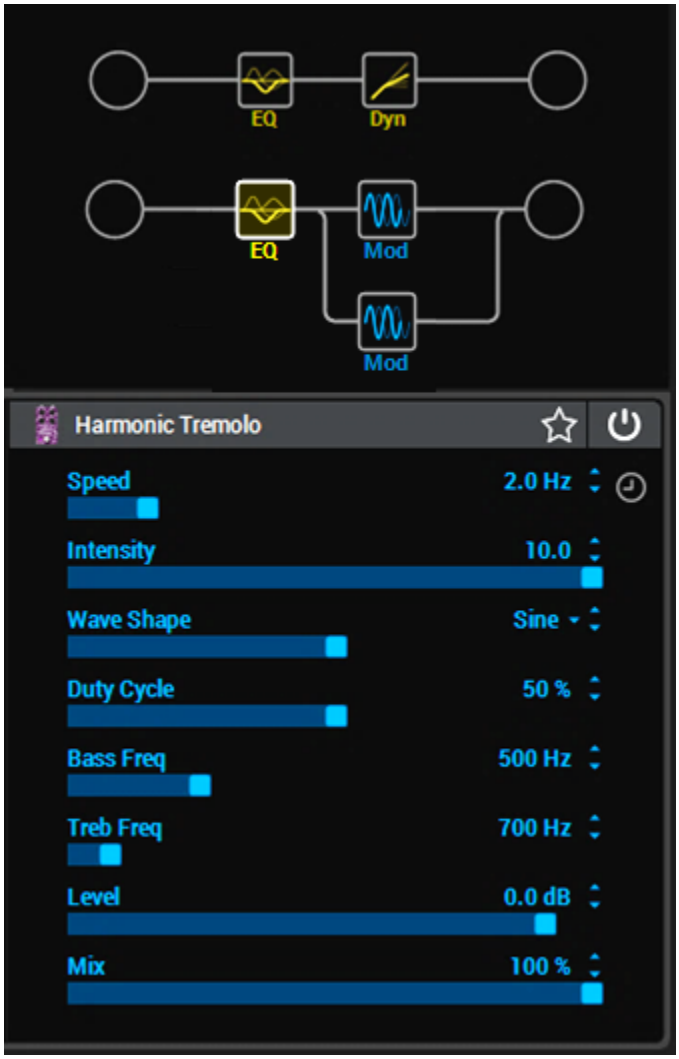


Figure 10.24 This preset bi-amps the bass. Path 2 applies stereo processing to only the high frequencies.

Path 1 is the straight bass sound. The Low and High Cut EQ filters out the highs above 1 kHz, and an LA Studio Comp smooths out the sound. Path 2 uses a Low and High Cut EQ to filter out the lows going to each sub-path. Each sub-path has its own Harmonic Tremolo processor.

Adding even a little bit of this effect can enliven the sound. Because the high frequencies are lower in level than the main bass sound, processing them is like adding a dusting of powdered sugar on top of a chocolate cake. In Path 2's Mixer, experiment with the balance of the two paths, and their panning.

Autofilter Articulation

B_3BArticulation is another 4-Path preset that processes the high frequencies. The highs split into two Autofilters (fig. 10.25). One's filter direction is Down, and the other filter's direction is Up. The intended effect is subtle, but the animation adds interest to the higher frequencies when layered with a solid low end.

Regarding the three LA Studio Comps, Path 1's compressor adds a little compression. The ones in Paths 2A and 2B apply more compression. They keep the filter resonance under control and maintain a consistent sound.

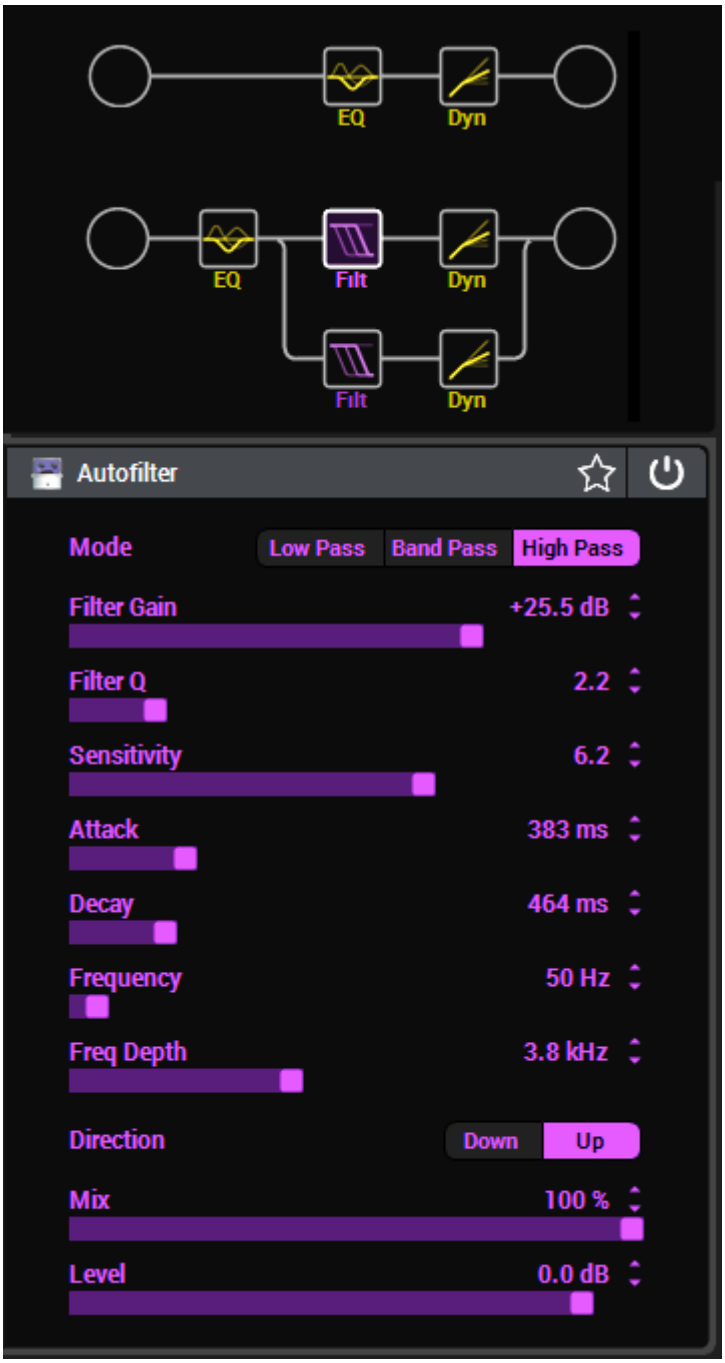


Figure 10.25 Bi-amping adds envelope-controlled articulation to only the higher frequencies.

Stereo Octave

The **B_3B StereoOctav** bass preset adds an octave-higher element but doesn't have an 8-string bass character. Instead, frequencies below 1 kHz provide the main bass sound in Path 1, while frequencies above 1 kHz feed the Input Split in Path 2 (fig. 10.26). Paths 2A and 2B transpose pitch up an octave, with a few cents of detuning. The Low and High Cut EQs in Path 2's sub-paths trim the highs, to prevent the octave-higher sound from being overbearing.

From there, each sub-path goes into a Double Take to diffuse the octave sound. This creates an "octave ambiance" superimposed on the main bass sound. Enough processing power remains to add an Amp+Cab in Path 1. You can even add an Amp+Cab in Path 2, although this seems less useful because you'll likely want to preserve the high frequencies.

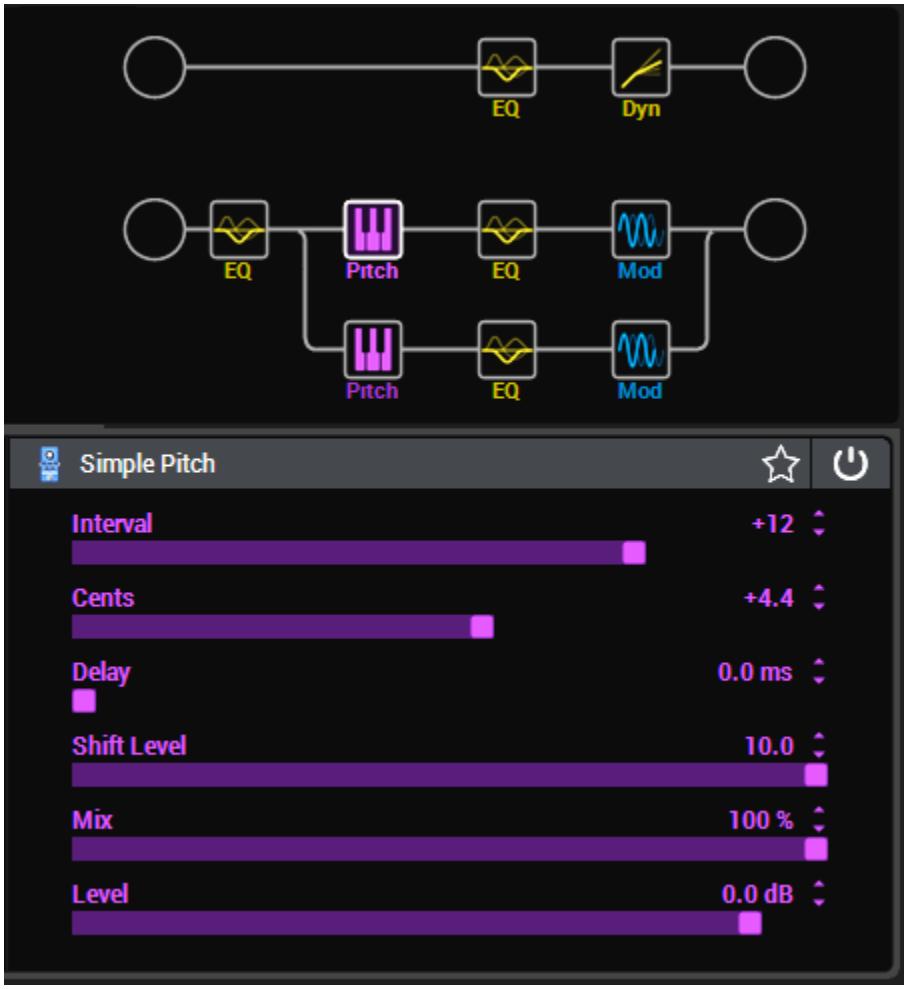


Figure 10.26 The octave-higher element adds presence to the bass.

The Path 2 Mixer's Level control is crucial for dialing in the right amount of the octave effect. Pull it down for a more subtle sound, but also try pushing it up to +6 dB—you'll hear a unique bass sound. It's a "ghostly" 8-string effect, because the octave-higher component isn't always prominent.

Chapter 11 | FAVORITES FOLDER FILES

The Favorites folder has a bunch of free files that I've found useful over the years—you might find them useful as well.

One of the great aspects of a downloadable eBook is being able to include presets and other files. This chapter is a grab bag of my Favorites presets. Some are simple, some are fun, some are utilitarian.

Unexpected Saturation

A few effects that aren't intended to provide saturation can be coaxed into providing it. For example, set the LA Studio Comp (fig 11.1, left) for no Peak Reduction, turn Gain up to increase saturation, then turn Level down to trim the output level. This tasty saturation effect is unique to the LA Studio Comp.

The Bitcrusher (fig. 11.1, right) gives overdrive effects. Choose settings that *don't* crush bits (e.g., 24-bit Depth, 44.1 Sample Rate) and *don't* gate the audio (e.g., set both Open and Close to -96.0 dB). Turn up the Gain. Mix can add in some clean sound. Low Cut and High Cut tailor the tone. As with the LA Studio Comp, adjust Level for the desired amount of level through the Bitcrusher.

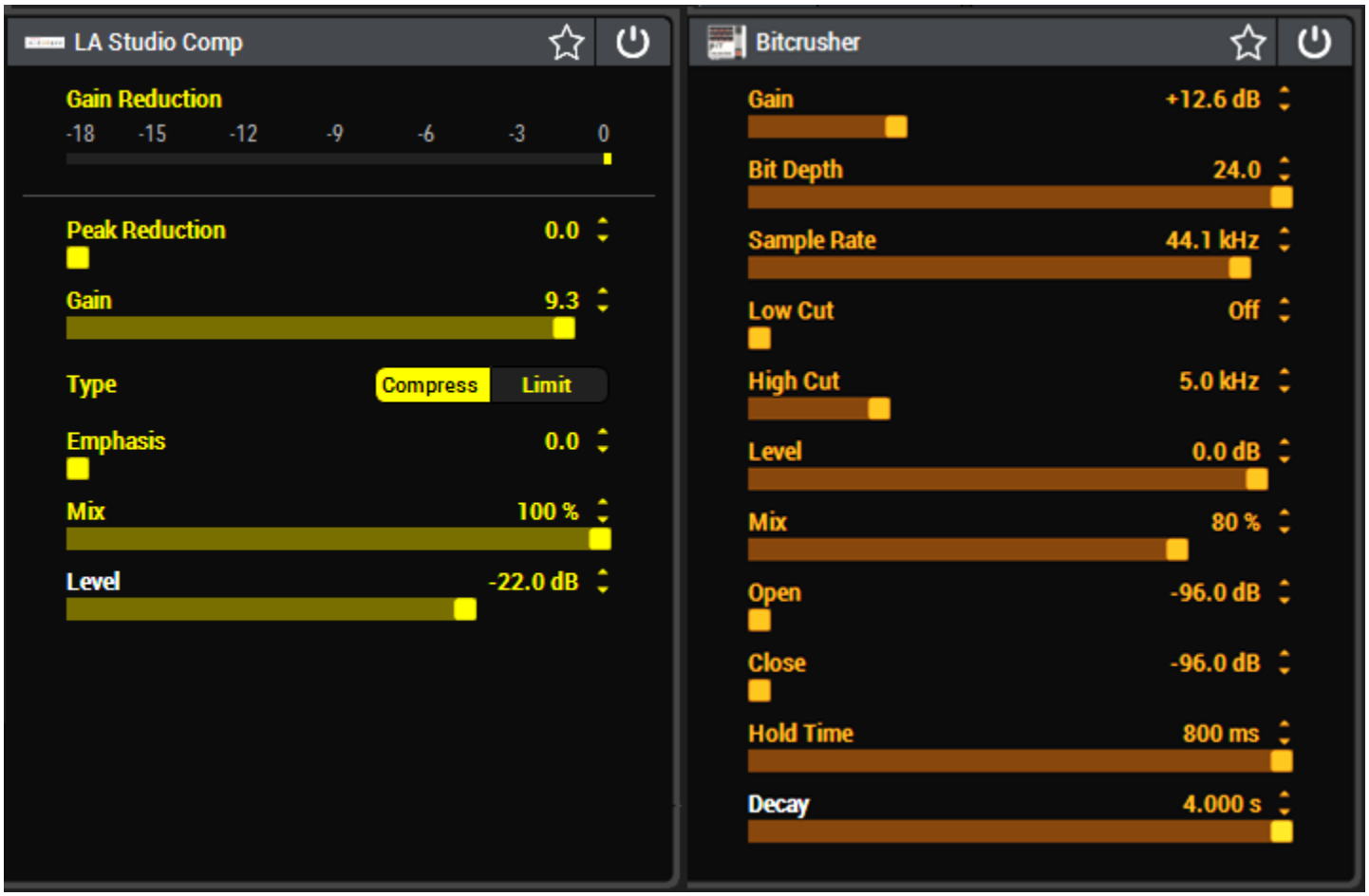


Figure 11.1 **LA Comp Sat.fav** and **Bitcrusher Sat.fav** provide saturation effects.

Targeting Equalizer

Helix has eight stock equalizers, but a ninth EQ is masquerading as a Dynamics processor. **TargetingEQ.fav** turns the 3-Band Comp into a a three-band graphic EQ that targets specific, adjustable frequency bands. Each band has it own level control.

To defeat compression so that the 3-Band Comp acts like an EQ, set all Thresholds to 0.0 (fig. 11.2). The Ratio, Attack, and Release settings don't matter, because the 3-Band Comp won't be compressing.

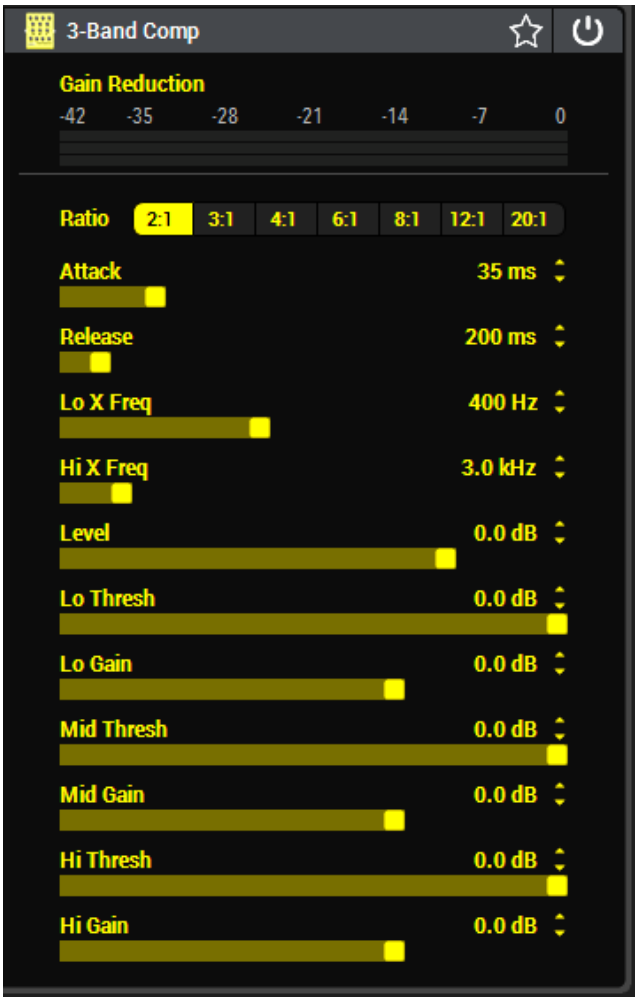


Figure 11.2 With all Thresh controls set to 0.0, the 3-Band Comp becomes a three-band equalizer.

To adjust the frequencies covered by the three bands:

1. To edit the lowest band, set Lo Gain to 0.0 and the Mid Gain and Hi Gain controls to -60.0 dB. Alter the Lo X Freq parameter to to set the highest frequency you want the low range to cover.
2. To audition the mid band, set Mid Gain to 0.0 and the Lo Gain and Hi Gain controls to -60.0 dB. Adjust the Hi X Freq parameter to set the highest frequency you want the mid range to cover. The Lo X Freq parameter sets the lowest frequency the mid range will cover.
3. Finally, set Hi Gain to 0.0 and the Lo Gain and Mid Gain controls to -60.0 dB. Check that Hi Gain covers the desired range of frequencies (you may need to re-tweak the Lo X and Hi X settings to optimize the sound).
4. Adjust the Lo, Mid, and Hi Gain controls for the desired balance of the three bands.

Tip: For Helix Native users, this is a versatile equalizer for many studio applications.

Dual Delay-Based Mono-to-Stereo Conversion

Mono2St_DualDela.fav converts a mono effects chain to stereo. It delays the mono signal by different amounts in the left and right channels. This creates stereo, because the two channels are different. The stereo effect is excellent in the studio, but this preset also collapses well to mono for live performance. There's no need to create separate presets or Snapshots for the studio and stage.

Although many Helix effects (like chorusing, or the Dual Cab option for amps) can generate stereo outputs from mono inputs, they're still effects. The Dual Delay gives natural-sounding stereo imaging effects with any mono input, including acoustic guitar.

The Dual Delay's Left Time and Right Time parameters have different, short values. To provide the stereo effect, turn up the Left Mix and Right Mix controls to around 40% (fig. 11.3).

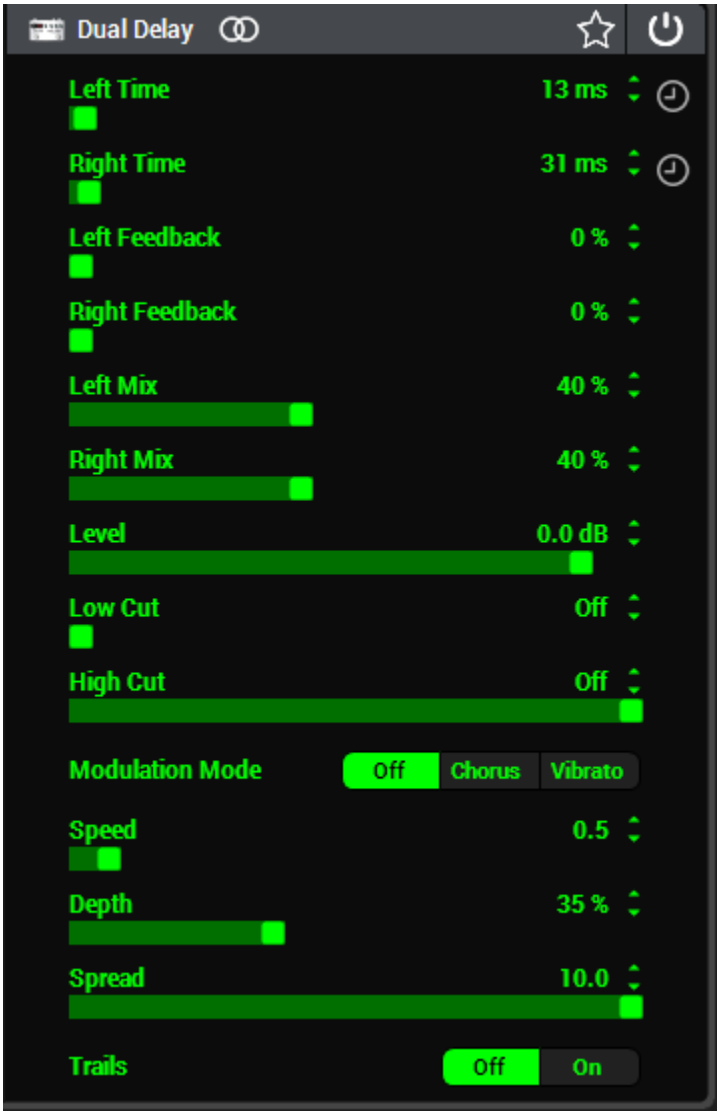


Figure 11.3 Time and Mix are the most important parameters for shaping the stereo image.

To help minimize resonances when mixing the two delays together, use prime numbers (5, 7, 9, 11, 13, 17, 19, 21, 23, 29, 31, 37, 41, or 43 ms) for the delay times. Too long a delay sounds like echo, so keep the longer of

the two delays under 37 ms. My preference is 13 ms and 31 ms, but also try 13 and 47 ms. The settings aren't too critical.

With delay-based stereo, the sound can become thin when collapsed to mono, unless you choose delay times that minimize cancellations. Also note that higher Mix control settings widen the image, but increase the potential for cancellation when summed back to mono. To check how the preset sounds in mono, insert a mono Vol block after the Dual Delay.

Use the following procedure to find the sweet spot of maximum stereo separation and minimal thinning when collapsed to mono:

- Start with the two Mix parameters set for 0%. Increase both parameters equally, until the sound becomes "phasey." Try different delay times to achieve the most natural sound, then reduce the Mix controls a bit to minimize any remaining cancellations. Settings of 30% to 40% work well. When you return the main output to stereo, you'll still hear a stereo spread.
- Increasing the difference between the two delay times often helps prevent thinning (and widens the stereo image, up to the point of sounding like an echo).
- Increasing the Feedback parameter values can add a hint of room-like ambiance. However, you probably won't want settings higher than 30% to 50%.

Tech Talk: Center-Channel Buildup

Creating a stereo image with delay often creates a thinner sound when played back in mono. One reason the delay-based technique used in this Favorite works well is because of a phenomenon called *center-channel buildup*, where left and right audio panned to center in stereo ends up 6 dB louder when collapsed to mono.

Because the left and right channels in this Favorite are a mix of dry and delayed sounds, the dry sounds become louder when the audio collapses to mono. This increases the dry signal level compared to the delayed signals, which emphasizes the sound's mono elements. With properly adjusted delay times and Mix parameters, stereo presets created for the studio can work well in mono for live performance.

For more about center-channel buildup, see the section on the Stereo Width block.

Simple Delay-Based Mono-to-Stereo Converter

Like the previous favorite, inserting the **Mono2St_SmplDela.fav** mono-to-stereo converter (fig. 11.4) at the end of a mono effects chain converts the output to stereo. Although this block has fewer editable parameters than the Dual Delay, it's still good at producing stereo effects that collapse well to mono.

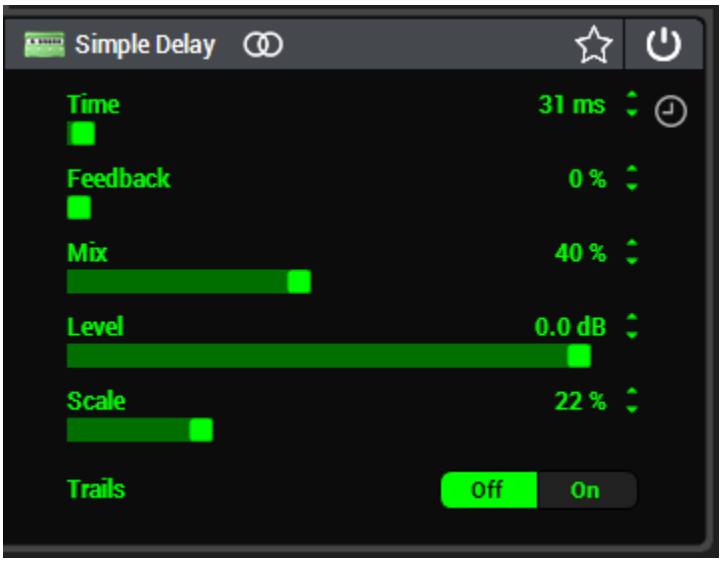


Figure 11.4 A stereo Simple Delay can add stereo imaging to mono presets.

Notes:

- Increasing the Mix makes for a wider stereo image, but also increases the risk of partial cancellation when summed to mono.
- Adding Feedback gives a hint of room ambiance. It's subtle, so you can push this as much as 50%.
- Compared to the previous mono-to-stereo converter based on the Dual Delay, the main advantage of this block is lower processing power.

Frequency Split Mono-to-Stereo Conversion

This stereo-to-mono conversion method for 4-Path Helix devices provides more subtle image enhancement and stereo spreading than the delay-based converters, and collapses well to mono.

As described in the Targeting Equalizer, defeating compression in the 3-Band Comp turns it into a three-band equalizer. Each band has an adjustable bandwidth. The Lo X Freq parameter sets the dividing line between the Lo and Mid bands. The Hi X Freq parameter sets the dividing line between the Mid and Hi bands. Different bands can then go to different locations in the stereo field. The Lo X and Hi X settings in fig. 11.5 are good defaults, but you may need to tweak them for specific applications.

Figure 11.5 All three 3-Band Comp blocks have the same parameter settings, except for the Lo, Mid, and Hi Gain levels. The left EQ passes high frequencies, the right EQ passes mid frequencies. The low frequency EQ isn't shown.

The **3-Band Comp.fav** blocks end up in three locations. Path 1 splits into two 3-Band Comp blocks. For Path 1A, set the Lo and Mid Gain to -60.0 dB, and the Hi Gain to 0.0 dB, so that only the high frequencies get through. For Path 1B, set the Lo and Hi Gain to -60.0 dB, and the Mid Gain to 0.0 dB, so that only the mid frequencies get through. At the Mixer, pan these oppositely. Higher panning values widen the stereo image. If the image seems unbalanced, trim the Pan parameters to compensate.

The third 3-Band Comp instance (its parameters aren't shown in fig. 11.5) goes in Path 2, and carries only the low frequencies. Set the Mid Gain and Hi Gain to -60.0 dB, and Lo to 0.0 dB. This doesn't require panning because normally, low frequencies are centered.

Tip: Tweak the levels to modify the sound further. For example, turn up the Hi Gain in the Hi band for a brighter sound, or the Mid Gain control in the mid band for more definition. Also, you can combine compression with equalization for dynamics control with stereo imaging.

Graphic EQ-Based Mono-to-Stereo Conversion

Inserting these blocks at the end of a parallel, mono effects chain, just before the Mixer, generates a stereo image that collapses extremely well to mono. Insert **Mono2St_GEQ 1.fav** at the end of one mono path, and **Mono2St_GEQ 2.fav** at the end of the other mono path (fig. 11.6). The 10 Band Graphic EQ band's sliders are set oppositely, so when one EQ is cutting, the other is boosting. Frequency response differences between the two paths create the stereo illusion.

At the output mixer, adjust the amount of separation with the A Pan and B Pan. The A Pan and B Pan needn't be panned full left and right. For example, if you want a guitar's image to extend from the left channel to center, set A Pan to Left 100, and B Pan to Center. However, because the stereo image collapses well back into mono, there's no technical reason (only artistic ones) not to pan the outputs to Left 100 and Right 100.

Figure 11.6 Two 10 Band Graphic equalizers can create a convincing stereo spread.

Tip: The Mixer's B Polarity parameter is set to Normal. However, also try Inverted. This cancels out any remaining audio the two EQs have in common, which creates a wider stereo spread. This still collapses well to a mono image.

However, the output level will be lower. Try both the Normal and Inverted options before committing to one setting or the other. Also, check the output in both mono and stereo, because the sound may "thin out" in mono if other effects are in the paths.

Pre-Amp Lead Fix

Insert **Pre-Amp Lead Fix.fav** (fig. 11.7) before an amp to help single-note leads stand out. Cutting the 31.25, 62.5, and 125 Hz bands takes out bass-heavy frequencies, which aren't needed for leads. If you're going to play much below the B string, push 250 Hz up to +10.0, dB and 125 to 0.0 dB.

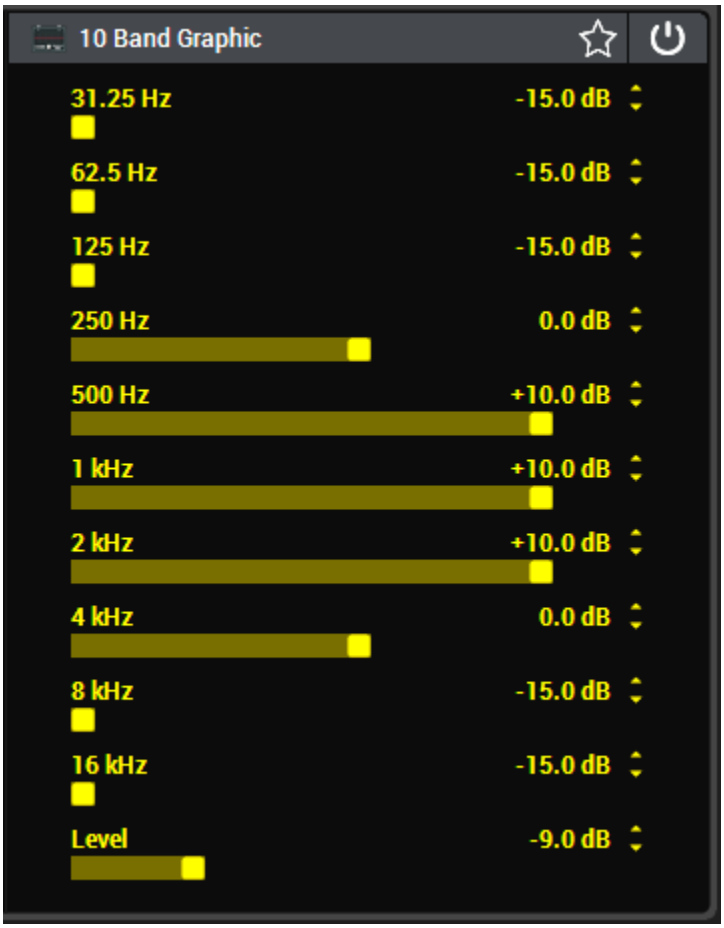


Figure 11.7 This 10 Band Graphic EQ setting "conditions" guitar for feeding sims.

The boosts at 500 Hz, 1 kHz, and 2 kHz "push" single-note leads into distortion before other notes. This makes sims feel more responsive and touch-sensitive. 4 kHz is pulled back to avoid harshness, and 8 kHz and 16 kHz are pulled all the way back because they don't contribute much to a meaty, yet bright, lead.

In general, adding EQ before an amp or other distortion-producing block doesn't influence tone as much as it tailors the sound for your particular touch and guitar. This technique also relates to compression. For more information, see the section EQ Before/After Compression.

Resonator

There are many ways to create a resonator aside from filters. The Multi Pass Delay (fig. 11.8) offers exceptional resonator potential. With short delay times, the various taps add a richness to the resonance that's not possible with other Helix effects. I wouldn't say this nails a resonator guitar effect, but it provides that feel. Three Favorites files cover three different keys: **Resonator_D.fav**, **Resonator_E.fav**, and **Resonator_G.fav**. However, the sense of pitch is broad enough that you can often use whatever sounds best.

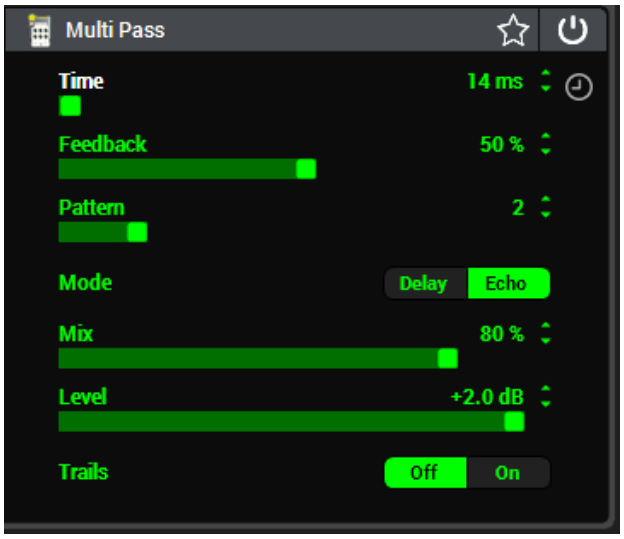


Figure 11.8 The Multi Pass Delay makes a fine resonator.

Set Mode to **Echo**. The resonance depends on the Pattern you've chosen, and the Time. Start with Pattern 2. The following delay times produce resonances that correspond to particular keys. Shorter times create a higher-pitched resonance effect, while longer times create a lower-pitched one. Although Helix can't specify tenths of a millisecond, the values shown are close enough to sound in tune.

- C4: 18 / 41 ms
- C#4: 17 / 39 ms
- D4: 16 / 36 ms
- D#4: 15 / 34
- E4: 14 / 32 ms
- F4: 13 / 30 ms
- F#4: 12 / 29 ms
- G4: 12 /27 ms
- G#4: 11 / 25 ms
- A4: 10 /24 ms
- A#4: 10 / 23 ms
- B4: 9 / 24 ms

Feedback and Mix determine the resonance's prominence. *Caution:* Keep the resonance below 90% to prevent self-oscillation and distortion.

Once you dial in the resonance for a particular key, alter the Pattern. The results can range from happy accidents to better resonance sounds. This effect won't sound exactly like a dobro…but with some editing, you can come surprisingly close.

Virtual Room

An acoustic space is a *physical* multi-effects processor that incorporates filtering, delay, and modulation. The **Virtual_Room.fav** block adds the short, subtle echoes that are characteristic of a small club or rehearsal space. In the studio, including these echoes will often help guitar fit well with other instruments that were miked in a room (drums, hand percussion, piano, etc.). This complements the Dynamic Ambiance effect already in Helix 3.50.

This effect isn't necessarily intended for live performance, because you're already playing in an acoustic space. Nonetheless, his effect can give a thicker sound with mono presets.

The Multitap 6's list of parameters is too long to fit comfortably on a single page, so fig. 11.9 breaks the interface into two halves. The left image is the interface's upper half.

Multitap 6
Time 31 ms
Feedback 20 %
Low Cut 100 Hz
High Cut 5.0 kHz
Mix 35 %
Level -1.0 dB
Tap 1 Scale 100 %
Tap 1 Pan Right 100
Tap 1 Level 10.0
Tap 2 Scale 93 %
Tap 2 Pan Left 100
Tap 2 Level 10.0
Tap 3 Scale 74 %
Tap 3 Pan Right 66
Tap 3 Level 10.0
Tap 4 Scale 61 %
Tap 4 Pan Right 40
Tap 4 Level 9.6
Tap 5 Scale 54 %
Tap 5 Pan Right 33
Tap 5 Level 10.0
Tap 6 Scale 42 %
Tap 6 Pan Left 33
Tap 6 Level 10.0
Trails Off On

Figure 11.9 These Multitap 6 delay settings create a virtual room.

Customization

- Longer times create a bigger virtual room (but too long a delay sounds less realistic).
- Although a little feedback creates a more complex ambiance, turning it up too far sounds metallic.

- The screenshot shows longer delays panned more to the sides, and shorter delays to the center. However, these are simply default settings.
- Bring the Mix parameter up just enough to fill in the virtual acoustic space. Too high a setting means the sound won't collapse well into mono.
- Mixing the shorter delay levels higher brings the guitar closer, while mixing the longer delay levels higher places the guitar further away.
- Adjust High Cut and Low Cut to taste. Cutting highs emulates a room with softer surfaces. Cutting lows reduces the potential for a "muddy" ambiance.
- A tap's Scale number is a percentage of the Time parameter (in this case, it's 31 ms). Using prime number delay times (13, 17, 19, 23, 29, and 31 ms) helps prevent echoes from resonating with each other. To convert a tap's Time to a percentage, divide the tap's target delay time by the Time parameter. For example, to obtain the 17 ms delay time for tap 4, divide 17 by 31. This gives 0.61, so set the scale parameter to 61%.
- Try inserting this effect after the German Mahadeva Amp+Cab. They get along well.

Multiband Utilities

Chapters 5, 7, and 9 cover multiband processing techniques. These insert EQ at a path's beginning to restrict the path's response to a specific frequency band. Options include 2-Band (bi-amping), 3-Band, 4-Band, and Combi-Band (3 bands in two paths, intended mostly for HX Stomp). Some techniques isolate bands better, while others have more overlap. They also require different amounts of processing power. See the Chapters linked above for more information. The five ways to create frequency bands in Helix are:

- The Input Split crossover for bi-amping
- 3-Band Compressor for two-, three-, and four-way crossovers
- 10 Band Graphic EQ (different sliders are higher or lower to create different frequency bands)
- Simple EQ
- Shelf EQ

With the last four options, to filter band-specific frequencies, simply insert the appropriate bands from Favorites at the beginning of the preset paths. Having separate blocks for different bands also makes it easy to mix and match splits. For example, blocks that provide the best isolation among bands tend to draw more processing power. Use these for paths that don't have a lot of effects. Use blocks that require less processing power for paths where you want to conserve processing power for other effects.

2-Band Splits for 4-Path Helix Devices

These allow Paths 1 and 2 of a 4-path Helix device to process low and high frequencies independently, thus allowing more effects than using an Input Splitter with two parallel sub-paths.

2B.HiSplitComp.fav Creates the high band with the 3-Band Comp.
2B.LoSplitComp.fav Creates the Low band with the 3-Band Comp.

Combi-Band Splits

Combi_3Comp_Mid.fav Creates a midrange band with the 3-Band Comp.
Combi_3Comp_Lo+H.fav Creates high+low bands by attenuating the 3-Band Comp's midrange.

Combi_Shelf_Mid.fav Creates a midrange band with the Low/High Shelf EQ.
Combi_Shelf_Lo+H.fav Creates high+low bands with the Low/High Shelf EQ by attenuating the midrange.

Combi_Simpl_Mid.fav Creates a midrange band with the Simple EQ.
Combi_Smpl_Lo+Hi.fav Creates high+low bands with the Simple EQ by attenuating the midrange.

Combi_GEQ_Mid.fav Creates a midrange band with the 10 Band Graphic EQ.
Combi_GEQ_Lo+Hi.fav Creates high+low bands with the 10 Band Graphic EQ by attenuating the midrange.

3-Band Splits (for 4-Path Helix only)

3Band_LoSplitGEQ.fav Creates a low band with the 10-Stage Graphic EQ.
3Band_MidSpliGEQ.fav Creates a midrange band with the 10-Stage Graphic EQ.
3Band_HiSpliGEQ.fav Creates a high band with the 10-Stage Graphic EQ.

3Bnd_LoSpltSimp.fav Creates a low band with the Simple EQ.
3Bnd_MidSpltSimp.fav Creates a midrange band with the Simple EQ.
3Bnd_HiSpltSimp.fav Creates a high band with the Simple EQ.

3Bd_LowSplitComp.fav Creates a low band with the 3-Band Compressor.
3Bd_MidSplitComp.fav Creates a midrange band with the 3-Band Compressor.
3Bd_HiSplitComp.fav Creates a high band with the 3-Band Compressor.

4-Band Splits (for 4-Path Helix only)

4B_LoSplitGEQ.fav Creates a low band with the 10-Stage Graphic EQ.
4B_LoMidSplitGEQ.fav Creates a low-midrange band with the 10-Stage Graphic EQ.
4B_HiMidSplitGEQ.fav Creates a high-midrange band with the 10-Stage Graphic EQ.
4B_HiSplitGEQ.fav Creates a high band with the 10-Stage Graphic EQ.

4Bd_LoSplitComp.fav Creates a low band with the 3-Band Compressor.
4Bd_LoMidSptComp.fav Creates a low-midrange band with the 3-Band Compressor.
4Bd_HiMidSptComp.fav Creates a high-midrange band with the 3-Band Compressor.
4Bd_HiSplitComp.fav Creates a high band with the 3-Band Compressor.

CA-X Eco Cabs

This collection of 10 cabs uses a single Parametric EQ block (fig. 11.10). Before version 3.50's efficient cabs, these could save DSP processing power. Although that's no longer the case, they still produce their own sound, and are easily customizable. Note that cabs using two parametric stages provide more nuanced cab emulations. See Chapter 6 for more information.

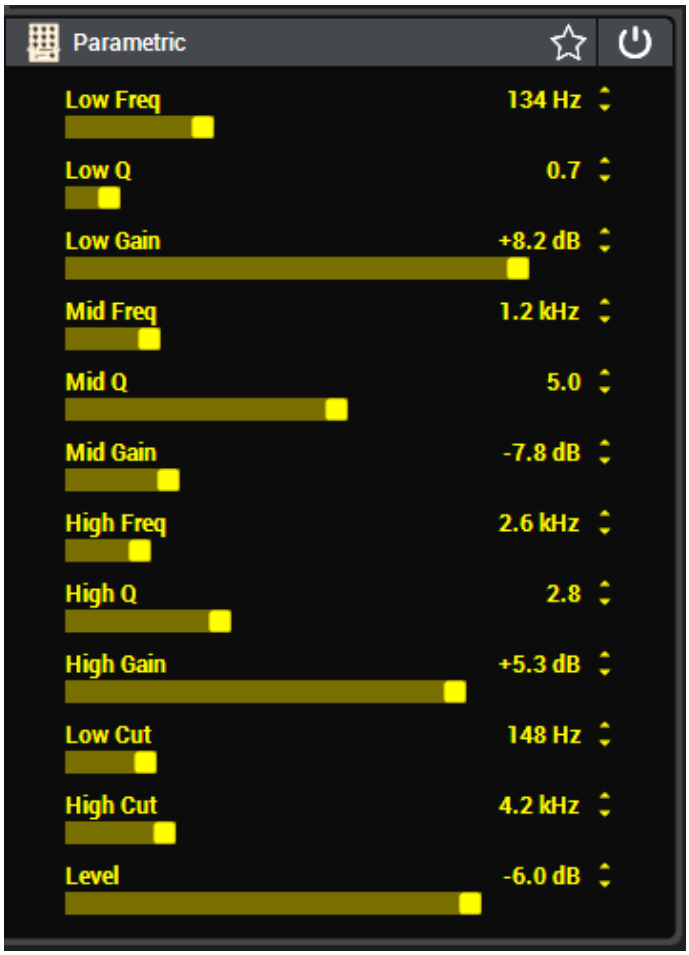


Figure 11.10 Typical Parametric settings for the **CA-X Eco 01.fav** EQ-based cab.

Including these as Favorites makes it easy to try these EQ-based cabs after an amp, so you can decide if any of them are helpful in giving a different kind of sound. You can also edit the EQ settings to tailor the sound more to your needs. The 10 cabs are:

CA-X Eco 01.fav
CA-X Eco 02.fav
CA-X Eco 03.fav
CA-X Eco 04.fav
CA-X Eco 05.fav
CA-X Eco 06.fav
CA-X Eco 07.fav
CA-X Eco 08.fav
CA-X Eco 09.fav
CA-X Eco 10.fav

Chapter 12 | THE OVERACHIEVING AUDIO INTERFACE

Helix is a cross-platform, high-quality audio interface—with features that even many dedicated audio interfaces don’t have.

Helix hardware devices are also cross-platform (macOS and Windows) audio interfaces. Although the interface features were optimized for Helix, this may be all the audio interface you need for your studio. The 4-Path Helix hardware is particularly capable. Standout features include:

- Monitor through effects with zero latency, while recording the processed sound and/or a dry version (for later processing).
- Use built-in options to re-amp guitar.
- HX Edit software can edit Helix hardware while it's serving as a USB interface.
- In addition to guitar, record a mic signal (4-Path devices only), keyboards, and other line-level devices.

The Helix manuals cover the interface functions well. This chapter supplements that information and covers some differences between 2-Path and 4-Path Helix hardware.

Helix 2-Path USB Inputs and Outputs

HX Stomp accepts three sets of USB outputs from the computer:

- USB 1/2 is the computer or DAW's main output. This goes to HX Stomp's L/Mono and Right outputs, and the headphones out. Initially, under *Global Settings* > *Ins/Outs* > *Page 2*, edit USB In 1/2 Trim to 0.0dB. You may need to turn this up or down later.
- USB 3/4 goes to the Send stereo output jack.
- USB 5/6 has no physical connection. It's an internal connection for re-amping, when the HX Stomp Input block is set to USB 5/6. This also allows using HX Stomp as a "hardware plug-in" for your computer, as described later.

HX Stomp can send six outputs to your computer. These can be:

- Six mono outputs
- Three stereo outputs
- A combination of the two (e.g., two stereo outs and two mono outs)

Here are common output configurations:

- USB 1/2 is typically a stereo output (with any processing) from Path A.
- USB 3/4 is typically a stereo output (with any processing) from Path B. Note that to send Path B to USB 3/4, in HX Edit you need to drag the output Mixer to Path B. This creates a Send to USB 3/4. You can still add a Send block in Path A.
- USB 5 and USB 6 provide mono outputs from the Input jacks (L/Mono and Right, respectively). The primary application is to provide a dry signal for re-amping. They can also send the output from a stereo keyboard or other instrument to your DAW.
- USB 7 and USB 8 provide mono outputs from the Return/Aux In jacks (L/Mono and Right, respectively). These are also useful for recording stereo instruments.

Tip: Some DAWs have an option to use external hardware the same way you'd use a plug-in, within a single audio channel. If your DAW can't do this, set a DAW track's computer output to USB 3/4, which sends audio to HX Stomp's Send stereo output jack. Patch the HX Stomp Send jack to the external hardware effect, and patch the external hardware effect's output to the HX Stomp Aux input(s). Insert a Send/Return block, with the corresponding Return/Aux In as its input. Insert a new track in your DAW, and set its input to USB 7/8. You can now record audio processed by an external hardware effect into this track.

To take advantage of Helix-based audio interfacing, you need to set up your DAW's audio routing to recognize the inputs and outputs. Fig. 12.1 shows Ableton Live's inputs, and what they receive from HX Stomp. Live's configuration menu can treat a stereo input as two independent mono inputs, if needed.

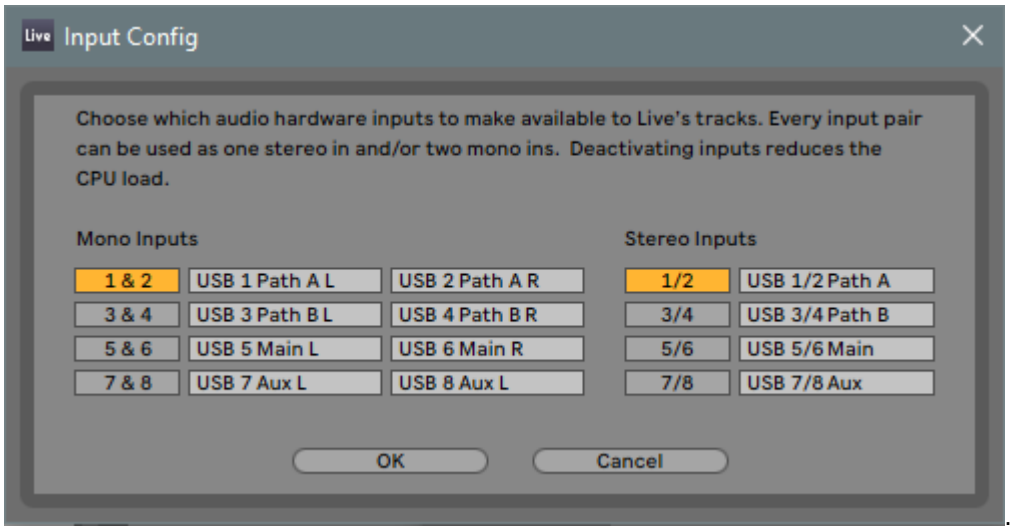

Input Config
Choose which audio hardware inputs to make available to Live's tracks. Every input pair can be used as one stereo in and/or two mono ins. Deactivating inputs reduces the CPU load.
Mono Inputs
1 & 2
USB 1 Path A L
USB 2 Path A R
3 & 4
USB 3 Path B L
USB 4 Path B R
5 & 6
USB 5 Main L
USB 6 Main R
7 & 8
USB 7 Aux L
USB 8 Aux L
Stereo Inputs
1/2
USB 1/2 Path A
3/4
USB 3/4 Path B
5/6
USB 5/6 Main
7/8
USB 7/8 Aux
OK
Cancel

Figure 12.1 Input setup for Ableton Live.

The way this is set up, any mono track input can select any HX Stomp mono output, and any stereo track input can select any HX Stomp stereo output. Other DAWs have similar setup options. Fig. 12.2 shows the equivalent input setup for Studio One.

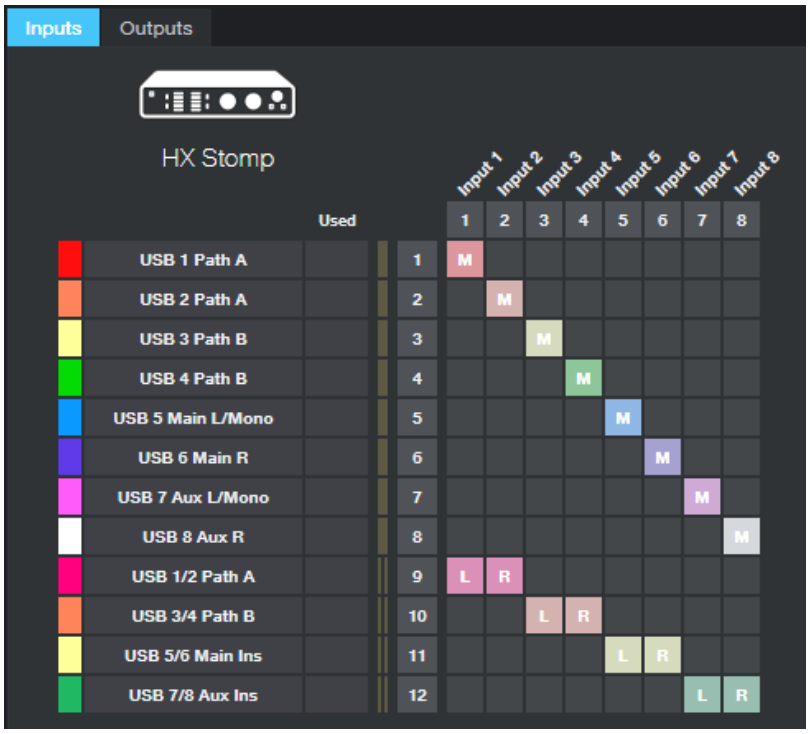

Inputs
Outputs
HX Stomp
Used
Input 1
Input 2
Input 3
Input 4
Input 5
Input 6
Input 7
Input 8
1 2 3 4 5 6 7 8
USB 1 Path A 1 M
USB 2 Path A 2 M
USB 3 Path B 3 M
USB 4 Path B 4 M
USB 5 Main L/Mono 5 M
USB 6 Main R 6 M
USB 7 Aux L/Mono 7 M
USB 8 Aux R 8 M
USB 1/2 Path A 9 L R
USB 3/4 Path B 10 L R
USB 5/6 Main Ins 11 L R
USB 7/8 Aux Ins 12 L R

Figure 12.2 Input assignments in Studio One.

Fig. 12.3 shows Cubase's output options. There's no USB 7/8 output, because HX Stomp reserves this USB pair for potential future use. Other DAWs have similar output configuration menus.

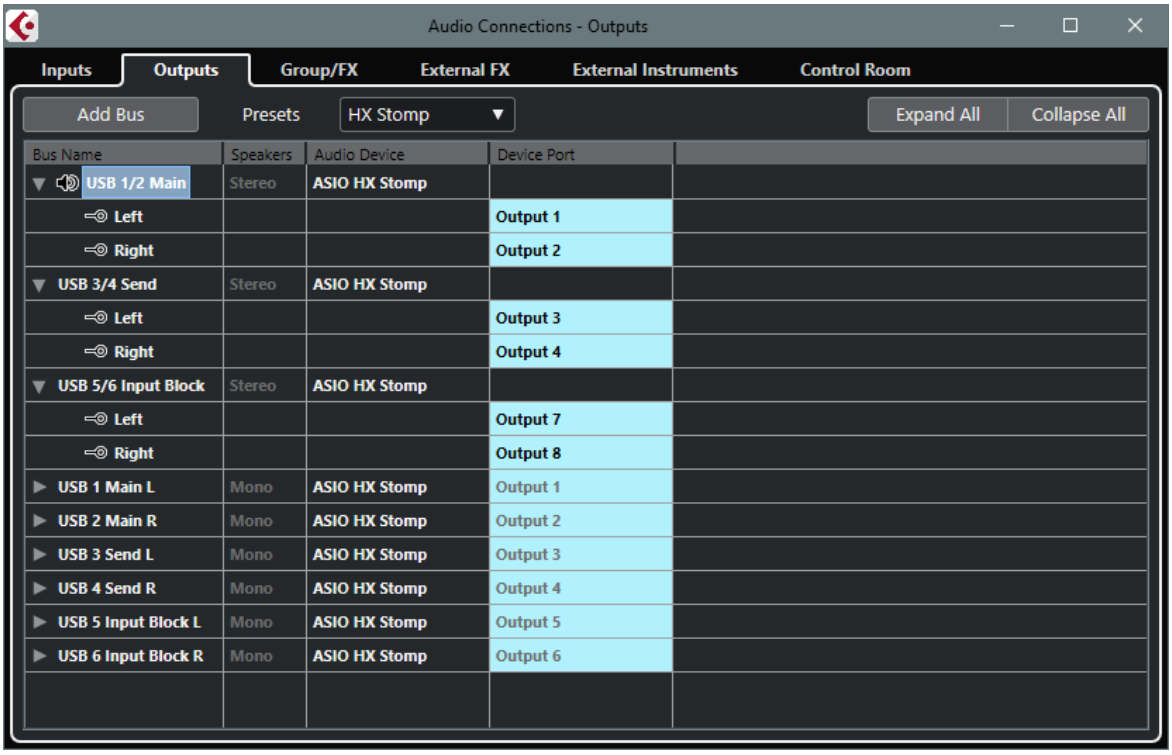


Figure 12.3 Output assignments in Cubase. The view shows the outputs used for each stereo bus.

Helix 4-Path USB Inputs and Outputs

4-Path devices like Helix Floor are a superset of the 2-Path devices. With DAWs, assigning 4-Path inputs and outputs is like assigning inputs and outputs in 2-Path devices, but with more I/O options. Here are the main differences between 4-Path and 2-Path devices.

Helix Floor accepts four sets of USB outputs from the computer:

- USB 1/2 is the computer or DAW's main output. This appears at the L/Mono and Right outputs, the XLR outputs, and the headphones.
- USB 3/4, 5/6, and 7/8 can accept computer outputs assigned to these buses. This flexibility not only accommodates re-amping, but also, sending different computer audio to Helix Path 1 and Path 2.

Helix Floor can send three stereo outputs and two mono outputs to your computer:

- USB 1/2, 3/4, and 5/6 can carry stereo outputs from Path 1, Path 2, and a Send block.
- USB 7 provides a dry mono output from the Guitar In jack.
- USB 8 provides a dry mono output from the Mic In jack.

There's a lot of routing flexibility! For example, with Helix Floor, you can make several simultaneous assignments at your DAW's track inputs:

- Assign a mono DAW track to USB 8 and record the mic signal.
- Assign a mono DAW track to USB 7 and record the dry guitar.
- Assign a stereo DAW track to USB 1/2 and record the Path 1 output.
- Assign a stereo DAW track to USB 3/4 and record the Path 2 output.
- Assign a stereo DAW track to USB 5/6 and record the audio from a Send block.

The following material emphasizes 4-Path hardware, but any functions also available in 2-Path hardware work similarly. Please refer to the Helix hardware Pilot's Guides for more information.

Audio Interface Overview

Most audio interfaces have an associated application for routing the inputs and outputs:

- Audio interface inputs appear as input options for a DAW's tracks.
- Audio interface outputs are available as destinations for a DAW's tracks and buses.

HX Edit provides a similar function. However, there are two main differences compared to typical interfaces:

- Helix Floor has a wide variety of input types—guitar, mic, Variax, Variax Magnetics, S/PDIF digital I/O, and Return jacks for send/receive jack pairs.
- Helix's heart and soul is hardware-based, real-time processing. Unlike traditional audio interfaces where zero-latency monitoring is an added feature, it's standard with Helix. You can record processed sounds into your DAW, and monitor the processed sound while recording, *in real time*. There's no latency, unless you monitor through the computer instead of monitoring through the Helix.

For example, Helix output USB 1/2 is a stereo output like you'd expect from any audio interface. However, it's a hardware-based output that's at the end of Helix's processing chain. So, as you monitor Helix's processed audio in real time, it's also being sent to your DAW.

Like most audio interfaces, Helix needs drivers. These link the Helix hardware with your computer. Drivers are available from the Line 6 website. Your choices for macOS and Windows drivers are similar to other interfaces. Make sure your drivers are up to date, to accommodate changes in your computer's operating system as well as in Helix.

Tip: Like any high-performance audio interface, do not use a USB hub, *especially when doing firmware updates.* To avoid possible connectivity issues, plug Helix into a USB port that connects to your computer's motherboard. Some people recommend using rear-panel USB connectors that are close to the power supply. Also, USB cables supplied with some consumer devices are used only for charging and do not carry data. *These cables will not work with any audio applications.* Use only USB cables intended for audio.

macOS and iOS

Helix is compliant with the macOS's Core Audio drivers. Patching a Mac's USB port to the Helix lists it as a selectable device in the Audio/MIDI Setup panel (located under Utilities). However, with the Mac's native drivers, only a 48 kHz sample rate is available. Downloading and installing Line 6's Core Audio driver allows choosing among 44.1, 48, 88.2, and 96 kHz sample rates, to match the rate selected in Helix.

iOS and iPhone devices need no drivers.

Windows

Helix works with Windows' native drivers, but download the Line 6 ASIO drivers for superior performance. Go to the Line6.com Downloads page. Under the All Software tab, choose HX Edit, your operating system, and click on Go. Download and install the latest HX Edit version, which also installs the needed drivers.

In your host program, open the Helix Control Panel (fig. 12.4). Your host may call it something else, like Settings. Or, it may have an entry under Preferences. The six ASIO Buffer Size choices are “Extra Small” to “Extra Large,” which correspond to 64, 128, 256, 512, 1024, and 2048 samples. Choose the smallest option that’s consistent with stable audio performance. While the window is open, set the Helix bit depth (24-bit is the usual choice). Choose Helix’s sample rate (44.1, 48, 88.2, or 96 kHz) in your host.

You can ignore the Sound Control Panel button, unless you want Helix to share playback with Windows’ native drivers (e.g., listen to YouTube from Windows while working with Helix). If so, click on Sound Control Panel, then the Speakers listed for Helix Audio. Make these Speakers the default device if it isn’t already. Click on its Properties, the Advanced Tab, and set the desired format for Windows playback.

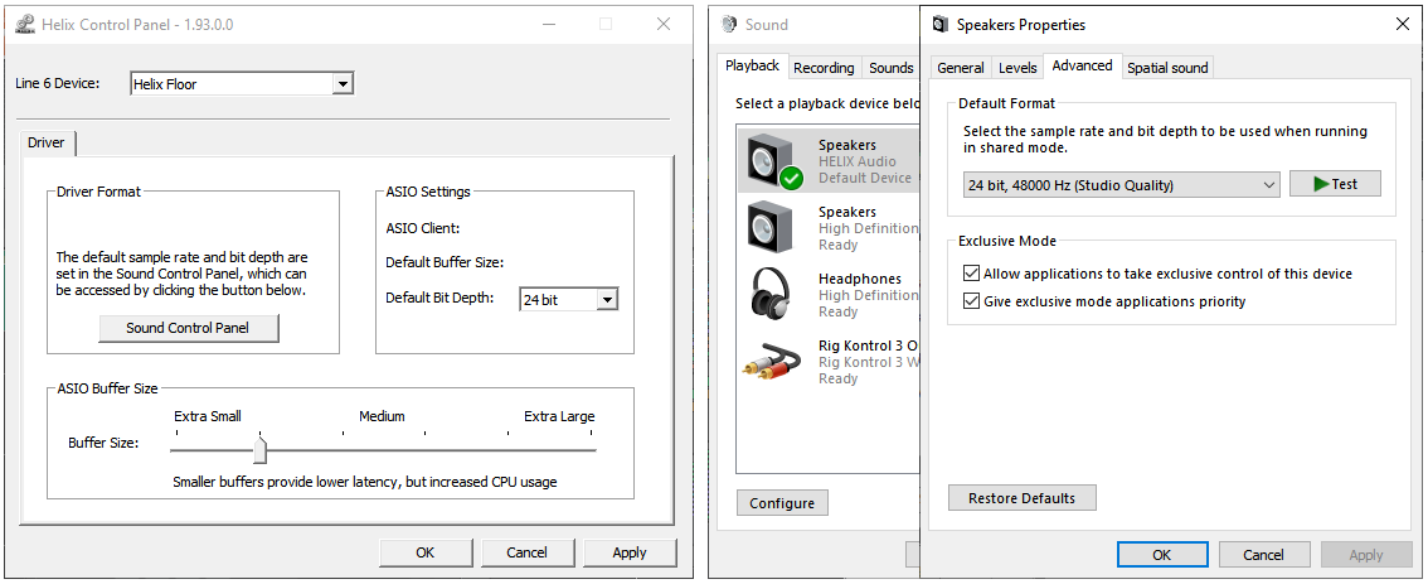


Figure 12.4 The Helix Control Panel is where you set up your computer to communicate with Helix. To share Helix playback with other Windows programs, also edit Windows’ native sound options.

Aggregating Interfaces

Helix is basically an 8-in, 8-out analog interface (there’s also S/PDIF digital I/O, and Variax inputs). If you need more I/O, you can add more inputs by *aggregating* multiple interfaces. This process integrates one or more audio interfaces to supplement what the Helix interface can do (for example, add several mic inputs for acoustic guitar projects).

macOS Aggregation

The Mac was designed to aggregate Core Audio interfaces. For example, suppose you have a Focusrite Scarlett 2i2, and later buy a Helix. Aggregation lets you use both interfaces as a team.

Open Audio MIDI Setup (under Utilities), and then choose the Audio Devices window (fig. 12.5). Click the little + sign in the lower left corner. An Aggregate Device box appears, and you’ll see a list of available I/O. Check the interfaces you want to aggregate, and check “Resample” or “Drift Correction” (depending on your OS version) for the secondary interface or interfaces. Now all input and output options from the aggregated interfaces will be available in your host program.

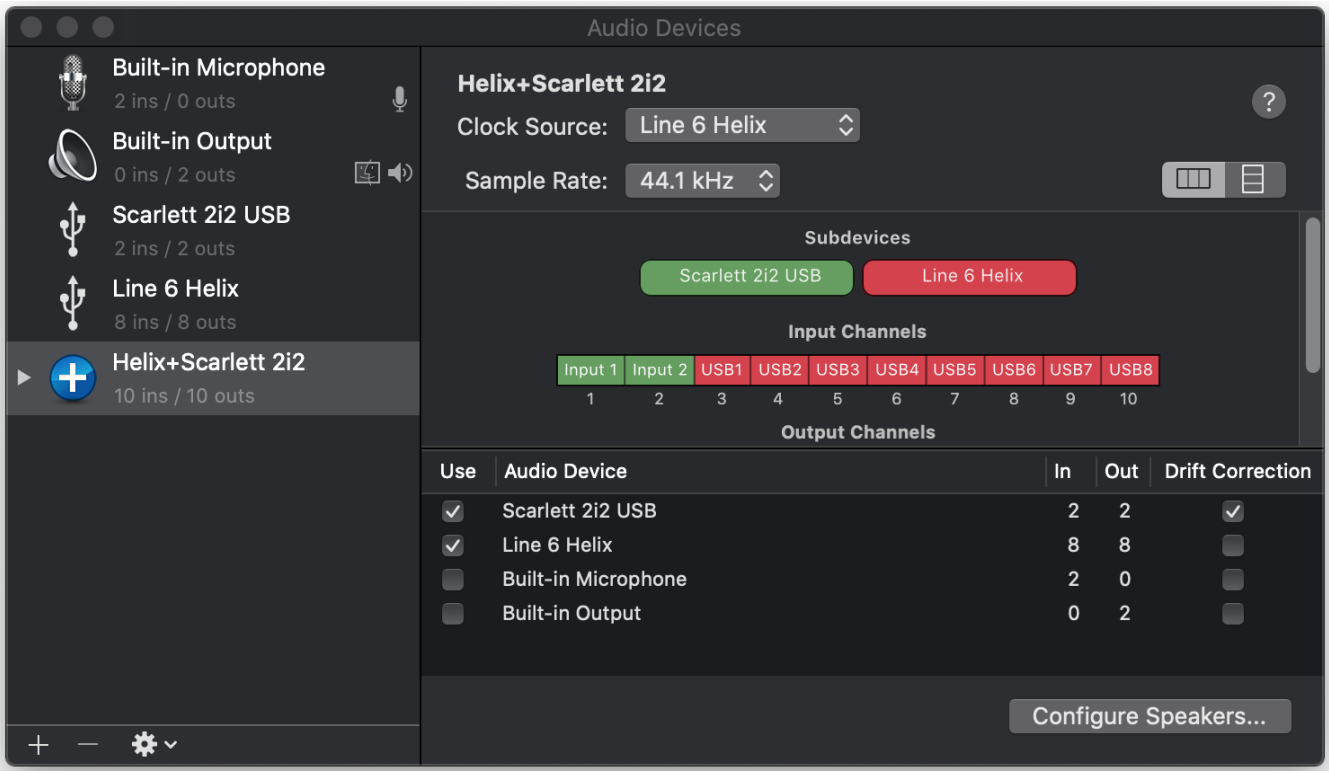


Figure 12.5 With the Mac, a Focusrite Scarlett 2i2 is supplementing Helix.

Windows Aggregation

Aggregating Windows devices is theoretically not possible with ASIO. However, you can use Windows' native drivers, like WDM/KS or WASAPI. Select one of these drivers in your host software, and all interfaces that support those drivers will show their available inputs and outputs in your application. Not all Windows programs support, or work well, with Windows' native drivers. At least currently, ASIO delivers better performance for Windows.

Some manufacturers (e.g., RME, Roland, PreSonus) get around not being able to aggregate ASIO by creating drivers that recognize their interfaces as one big interface instead of multiple interfaces. However, this has constraints. The interfaces have to be at least from the same manufacturer, and may only work with interfaces that share common port protocols (e.g., Thunderbolt or FireWire).

Regardless of how you choose to add more inputs, having enough inputs can serve as a patchbay you never have to re-patch, because everything you want to record will be only a couple of clicks away.

Helix Global Settings and Audio Interfacing

HX Edit doesn't expose all of the Helix parameters that relate to audio interfacing (like choosing line or instrument level). You'll need to call up a few of them on the unit itself, under Global Settings. These parameters don't necessarily need frequent adjustment, but they are key to setting up the I/O.

There are *many* ways to set up Helix for different studio and live performance scenarios, and to cover all scenarios would take a book of its own. So, we'll cover the main aspects of using Helix as an audio interface for computer applications, and the associated Global settings.

HX Edit Software

Although you can edit what we're about to describe in the Helix floor unit, since you're using it as a computer interface, take advantage of the HX Edit program. You set up the I/O in the Input and Output blocks. Some of the routings may seem confusing, because they're intended to offer creative possibilities as well as utilitarian ones. However, the Multi Input and Multi Output blocks will cover most of your needs. Matters get more complex only when using Helix to go beyond simply getting audio in and out of a computer.

Multi In and Out, USB 1/2 Destination

In Helix Floor's Global settings, on page 1 assign USB 1/2 as the Multi Destination. Multi In listens to anything that looks like a guitar: Guitar, Aux, and Variax. The adjacent USB 1/2 Trim control sets the level. The Multi Out is post-Helix processing, which plays back through XLR, ¼", headphones, and USB 1.2 outputs.

Remember that the Multi blocks are part of the Helix hardware, so they're independent of whatever is happening within your computer. Also, all your DAW's tracks play back through USB 1/2.

Recording with a Mic

Helix can record vocals (and do tasty vocal processing—it's a well-kept secret). Plug the mic into the rear-panel Mic input. Global Settings for Mic include 48V Phantom power on/off for condenser mics, Mic In gain (up to 60 dB), and Mic In Low Cut. Adjust these for the best performance with your mic.

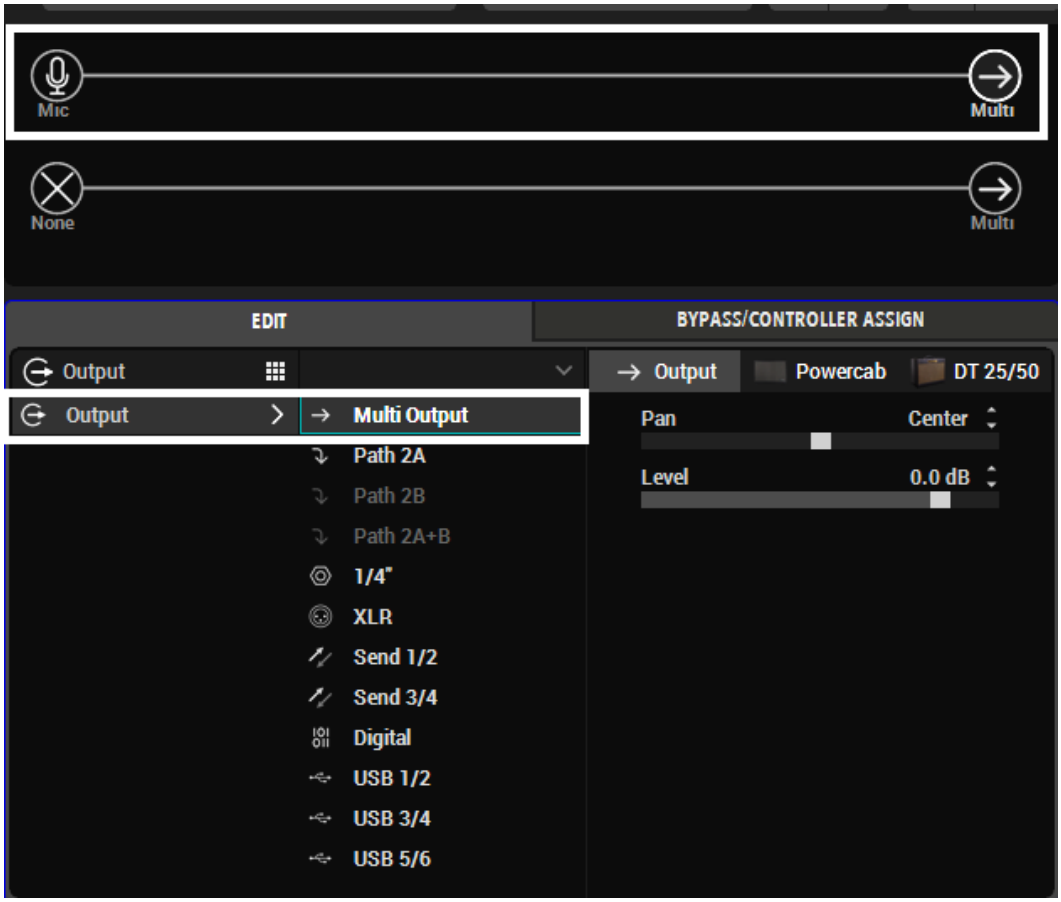


Figure 12.6 The top section (outlined in white) shows the Mic Input block feeding the Multi output block. The outlined section in the middle confirms that Multi Output is selected.

1. Choose Mic for the Helix input block (fig. 12.6). Choose Multi for the output block, and you'll hear the mic with zero-latency monitoring. Play back your DAW, and you'll hear the mic mixed in with the DAW's outputs.
2. To monitor the mic with Helix effects (like compression, EQ, reverb, etc.—whatever the vocalist likes), insert the processor(s) between the Mic input and the Multi output.
3. To record the processed track in mono, assign a mono DAW track's input to Helix Input 1 or 2. For stereo effects, assign a DAW's stereo track to Inputs 1+2.
4. Enable record, and you'll record the processed mic sound. (Conceptually, this recalls Universal Audio's powered plug-ins, where you can monitor with effects *and* record the effects, with zero latency.)

Along with USB 1/2, stereo streams are also available from USB 3/4 and USB 5/6. These can be treated as two independent mono streams. Even with the mono mic input, if you insert a mono in/stereo out or stereo processor into Helix, the output will be stereo. (Note that the following examples use USB 3/4, but you can substitute USB 5/6 instead—they're functionally the same.) You can simultaneously record the dry sound, while monitoring with effects at zero latency (fig. 12.7). This gives the benefits of hearing effects while recording, with the dry signal available for subsequent processing. Think of it as re-amping for vocals:

1. Set both input blocks to Mic.
2. To record the mic signal without any added effects, send the dry Mic path to USB 3/4. This feeds the mono mic signal to both USB 3 and 4, so assign a mono DAW track input that's recording the mic to either input 3 or input 4, or set a stereo DAW track to inputs 3+4. Do *not* enable the DAW track's input monitor.
3. To also monitor with effects, set the second path to Mic, insert an effect, and send its output to Multi. You'll hear the processed sound mixed in with your DAW's tracks, while the track into which you're feeding the mic records the dry sound.

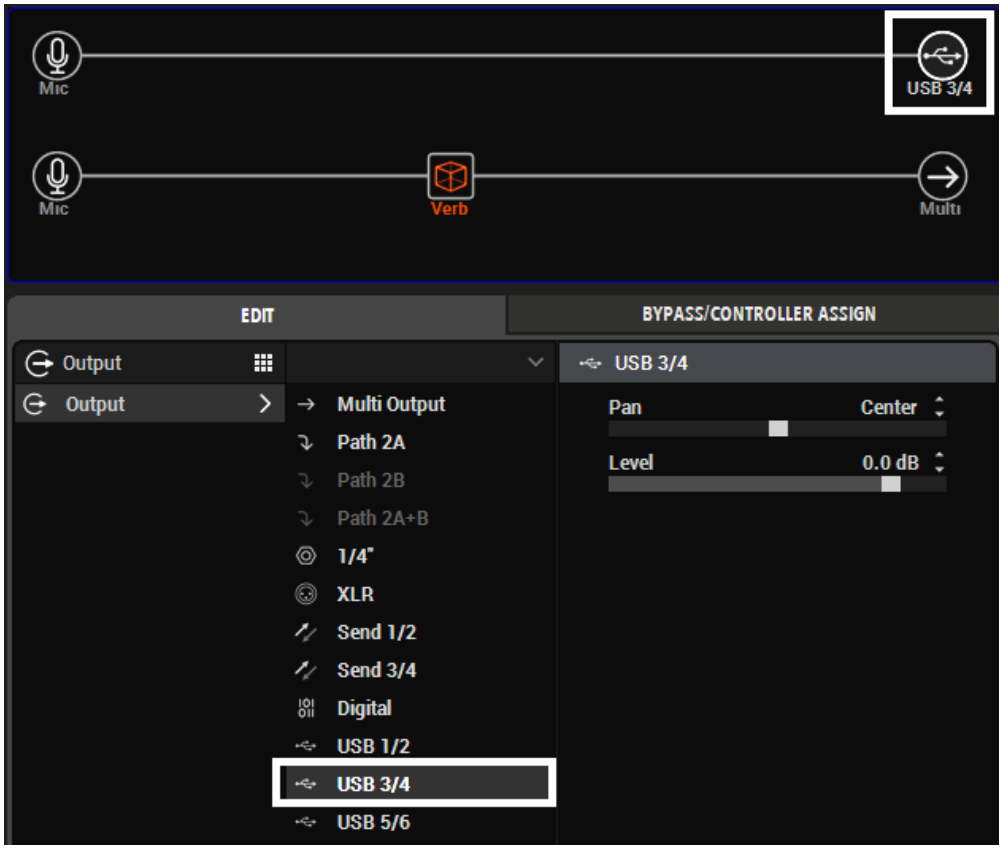


Figure 12.7 Sending the mic to USB 3/4 (outlined in white) allows recording it dry. Creating an additional path to send the mic through effects, and routing it to the Multi out, lets you monitor through effects with no latency. With this setup the processed audio is not being recorded, but simply combined with what you hear from the DAW's tracks for zero-latency monitoring through effects.

Here's another variation. Suppose you want to monitor through plug-in effects inserted in your host's virtual console, instead of the Helix effects. Do this by sending the mic signal directly to a USB bus output, and setting the DAW's track input to this output:

1. Similarly to fig. 12.7 above, send the mic to USB 3/4.
2. In the DAW track where you want to record the mic, set the input to record Input 3 or 4 for a mono track, or to Inputs 3+4 for a stereo track.
3. Enable the track's input monitor.
4. Set the second Helix path's Input to None.

Now the mic bypasses Helix's processing and goes directly to the computer. Since you're monitoring the computer's output, you'll hear any effect plug-ins inserted in a DAW's track. However, now that you're out of Helix's hardware environment and monitoring through a computer, there will be latency.

Recording Guitar

Similar techniques to recording a mic apply to guitar, whether this is a guitar plugged into the input jack, aux input (e.g., for active guitars with super-hot pickups), or a Variax plugged into the Variax input. Select the desired input option with the Input block. As with the mic examples, you can monitor through Helix effects with zero latency, while recording the processed or dry sound.

Recording Line-Level Instruments

To record a mono or stereo drum machine, synthesizer, etc., the effect loop Receive jacks can also serve as inputs (fig. 12.8). To use these:

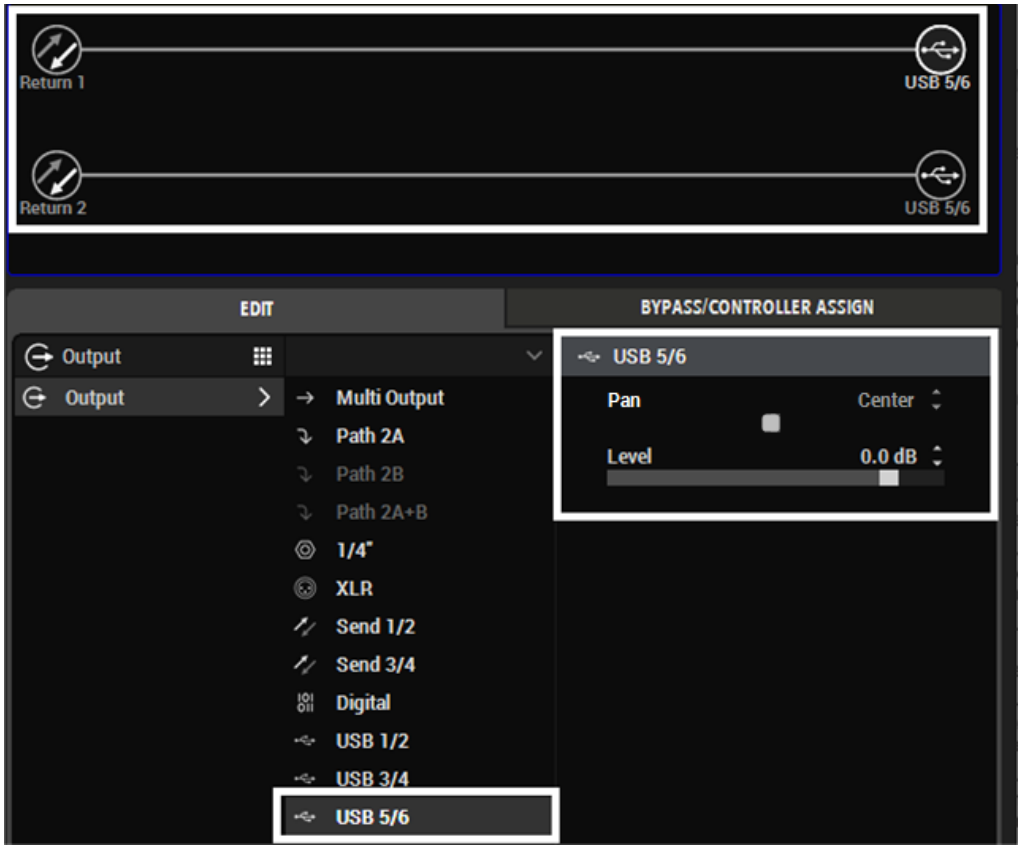


Figure 12.8 Setup for recording a stereo signal, like from a drum machine or keyboard.

1. For stereo instruments, patch into two of the loop Receive jack inputs (1 and 2, or 3 and 4—not to be confused with the similarly numbered USB I/O).
2. Under the Global Settings, choose whether these are Line- or Instrument-Level inputs (this also sets levels for the associated Send jacks).
3. Set the input blocks to Return 1 and Return 2 (or 3 and 4). The output blocks go to USB 3/4 or USB 5/6.

Unlike mono signals, we can't record the inputs dry and monitor with effects, although as described above, you can record with effects inserted between the Input and Output blocks, as well as monitor through computer plug-ins.

What happens at your DAW depends on how it does input routing. With mono tracks, assign one track's input to Helix Input 5, the other to Helix Input 6, and pan them oppositely. With a stereo track, assign its input to Helix Inputs 5+6.

Tip: Helix also has 5-pin DIN MIDI-compatible connectors, so it's a MIDI interface as well as an audio interface. Plugging a keyboard controller's MIDI out into the Helix's MIDI in can control virtual instruments that are loaded in your computer.

The DI Shortcut

So far, we've covered how to take advantage of the Helix hardware to provide zero-latency monitoring, even with processed sounds (which we can also record). However, Helix has a special "shortcut" for recording a DI signal. Inputs 7 and 8 tap off the input signals before they're processed by Helix, so recording into a stereo DAW track assigned to inputs 7+8 (or a mono track assigned to input 7 or 8) preserves the dry audio for later re-amping. In the Global Settings for Ins/Outs, you can assign inputs 7 and 8 independently to guitar, mic, aux, Variax, or Variax magnetic pickups.

This means you can mic a physical amp, and record that with or without effects into a track, while also recording the dry signal into another track. Or, record direct guitar through Helix effects into a track, while also recording the dry signal into another track.

Re-Amping with Helix

The easiest way to re-amp tracks recorded dry is to use the Helix Native plug-in. It has the same processors as hardware Helix devices, and you can exchange presets between the floor unit and Helix Native software. Just process the dry track with Helix Native as an insert plug-in—instant re-amping.

If you don't have Helix Native, most DAWs can also re-amp though Helix (fig. 12.9).

1. You'll need two tracks in your DAW. Record your dry guitar, or if you recorded the input from USB 7/8 into a track, use that for your dry track.
2. Assign the DAW track's output, or a send from the track, to USB 3/4 or (USB 5/6).
3. Assign a Helix Input block to USB 3/4 or (USB 5/6).
4. Now the Helix input is listening to the dry track, and you can insert Helix effects blocks to process the sound.
5. Assign the Helix Output block to USB 1+2.

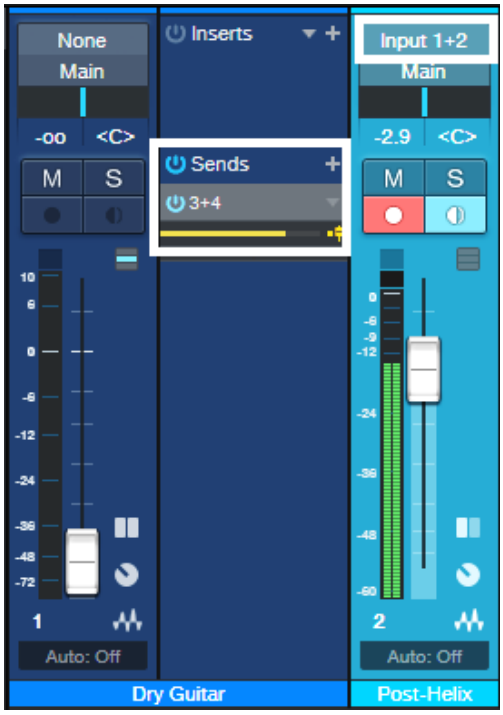


Figure 12.9 The Dry guitar is sending its audio to USB 3/4; Helix's Input block is set to USB 3/4. Helix processes the signal, which goes to the Multi Output block. Its audio appears over USB 1/2. Setting another track's input to USB 1/2, and recording on that track, captures the re-amped sound.

6. Because the processed Helix sound feeds USB 1+2, you'll hear the processed output in your headphones or monitor speakers.
7. After obtaining the sound you want, add another track in your DAW, and set its input to Input 1+2. Record-enable the track, start recording, and you'll record the processed Helix sound.

Using HX Stomp as a Pseudo-Plug-In

HX Stomp can re-amp similarly to Helix Floor. The Pilot's Guide documents the procedure. If you don't have Helix Native, re-amping allows HX Stomp hardware to act like a plug-in for your DAW. There are two main limitations:

- You can hear only one instance in real time when going through hardware. Helix Native software allows as many instances as your computer can handle.
- To use more than one instance, you need to bounce (record) the processed sound into a different track. This frees up Helix for use in another track.

With Windows, this process assumes HX Stomp is your sole ASIO audio interface. With the Mac, you can aggregate it with an additional interface.

1. Set the outputs for any DAW tracks you want to monitor to USB 1/2. You may need to check that the DAW's master fader in its virtual console is turned up. Test playback to verify that you can hear the song's tracks.
2. Set the HX Stomp input block to USB 5/6.
3. Note the DAW source track you want to bounce through Helix. Set its output to USB 5/6.
4. Insert a new track. Set its input and output to USB 1/2 and choose its Record mode.
5. It's good practice to turn down the new track's channel fader in your DAW's console. You'll likely need to enable the track's input monitor function (e.g., a small button with a speaker in the channel strip).
6. Start recording. The audio processed by Helix will be recorded into the track you inserted in step 4.

7. To play back the processed track, mute the DAW’s source track. Don’t erase the source track yet, in case you need to re-do this bouncing procedure later on.
8. Raise the level for the track you recorded in step 4, and you should hear the bounced track.

The Bottom Line

Most guitar players would describe Helix as a really cool multi-effects. That’s true. But it’s also an audio interface, and the same effects used for guitar can apply to voice, keyboards, and more. You can even monitor through effects with zero latency, as well as do re-amping.

Sure, you can’t set up four mics and record a drum set. But for solo and duo artists, having a Helix may eliminate the need for a separate audio interface.

APPENDIX

HELIX
01B Dream Rig Duet
06B Seismic Matters
HX STOMP
EXP 1 EXP 2
PHONES
BYPASS
ACTION
PAGE
MODE
HOLD TO EDIT
Harmony Delay
Ubiquitous Vibe
Mystery Filter
Stream 808
TAP
01D DAW Remote
01C Jimi, Not Jimmy
01B Dream Rig Duet
01A None More Black
BANK
Delay Harmony Delay
Time
Feedback
Key
Scale
Mix
Level
Minor
40%
0.0 dB

Those who have been using Helix for a long time may not be aware of some of the more recent updates—which go beyond just new amps or effects. So, this Appendix consolidates material issued previously in the release notes for various versions, other than material already covered in the book. Thanks to Line 6 for permission to include portions of their release notes in this eBook.

3.01 Update

True Preset Spillover

(Helix Floor, Helix Rack/Control, Helix LT)

Despite what some might have you believe, the only digital multi-effects units with *true* preset spillover were the DigiTech GSP1101, 2112, and 2120, and they accomplished this by dedicating one of their two DSPs to spillover and only spillover. (To hear two dynamically allocated presets with absolutely zero gap—not a global reverb smeared to cover the gap—you need enough DSP to run both of them simultaneously.)

If you're willing to sacrifice half your DSP by *completely disabling path 2*, you can have true preset spillover. If you don't want to lose Path 2 just for spillover, then keep using snapshots. To enable preset spillover:

1. *Global Settings* > *Preferences* menu, set Preset Spillover to **On**. (Shortcut: Hold **ACTION** and press **HOME.)** A dialog appears, reading "Remove Path 2 to enable preset spillover?"
2. Press **Knob 6 (OK).**
3. Press **HOME.** Path 2 has disappeared! You may now switch presets to your heart's delight with true spillover. However, note that if Preset A is still spilling over into Preset B (for example, one of its delay blocks' feedback is making the delay self-oscillate), switching to Preset C will abruptly cut off Preset A.

IMPORTANT! Unfortunately, Looper recording or playback will *not* currently spill over from one preset to another. Also, note that there may be a slight hiccup when switching between two presets with different impedance values for the Guitar Input.

To return to normal operation with two paths:

1. Turn **Preset Spillover** back to **Off.** (*Shortcut:* Hold **ACTION** and press **HOME** again.) A dialog appears, reading "The preset must be reloaded. All unsaved changes will be lost!" So, if you've made changes to the preset, be sure to save it before turning Spillover off.
2. Press **Knob 6 (OK).**

Favorites

(Helix Floor, Helix Rack/Control, Helix LT, Helix Native, HX Effects, HX Stomp)

If you've dialed in an amp or effects block exactly the way you like it, you can now save its settings as a favorite so it can be added very quickly, complete with bypass state and footswitch assignment. Favorites appear in a new "Favorites" category and can be renamed, reordered, and cleared.

Tip: If you don't want a favorite to be recalled with its footswitch assignment, add the block before you assign a switch.

To create a favorite:
1. Choose one of your favorite amps or effects.
2. Tweak it exactly how you like it, with or without footswitch assignment.
3. Press **ACTION**, and then press **Add to Favorites.** The Favorites list appears.

4. If you want to replace an existing favorite (there are128 total), select it and press **Replace Favorite**. If you want to add a new favorite, select the location where you want to place it, and press **Add New Favorite**.

To modify the Favorites list while it's open, press **ACTION.** Then:

To reorder the selected favorite up or down the list, turn **Reorder Favorite** (on HX Effects, press **Reorder Up** or **Reorder Down**)
To rename the selected favorite, press **Rename Favorite**.
To clear the selected favorite, press **Clear Favorite**.
To clear all favorites, press **Clear All Favorites**. A dialog appears. Press **OK** to confirm.

To call up a favorite:

At any time, select an empty block and turn the joystick (**Big Knob** on HX Effects or **Lower Knob** on HX Stomp). All your perfectly tweaked favorite amps and effects instantly appear, without having to open the model list at all. If you typically keep model categories on the same stomp footswitches, you could conceivably create a new preset with all your favorite stuff—including all footswitch assignments—in less than 10 seconds. Creating tones has never been faster or easier.

Tip: If you keep one or more external pedals connected to Helix's FX Loops, add the FX Loop blocks as favorites and then rename them after the real thing.

User Model Defaults

(Helix Floor, Helix Rack/Control, Helix LT, Helix Native, HX Effects, HX Stomp)

You can save any amp, cab, or effect block's settings as a user default. So, every time you call up that model, it sounds exactly the way you want it. If you want to revert it to the factory default, you can do that as well.
NOTE: Unlike Favorites, user model defaults do not store footswitch assignments. To create a user favorite:

1. Choose any model and tweak it exactly how you like it.
2. Press **ACTION** and then **User Default.** From Helix Native or *HX Edit*, right-click (Mac: control-click) the block icon and select "User Default."

Auto Impedance Preference

(Helix Floor, Helix Rack/Control, Helix LT, HX Stomp)

A new ***Global Settings > Preferences > Auto Impedance*** parameter determines how the Guitar In's impedance circuit behaves when *Input > In-Z* is set to **Auto**. When set to **First Block** (the default), the impedance circuit reflects the impedance of the first block on Path 1A, regardless of whether it's enabled or bypassed. When set to **First Enabled**, the impedance circuit reflects the impedance of the first *enabled* block on Path 1A.

Min/Max Value Indicators

(Helix Floor, Helix Rack/Control, Helix LT, HX Stomp; feature already in HX Effects)

Parameters assigned to controllers or snapshots now display small **Min Value** (below) and **Max Value** (above) indicators. You can now see the range of continuous controllers from the **Home** screen.

8 Blocks in HX Stomp

(HX Stomp)

HX Stomp's simultaneous block count has been increased from six to eight.

IMPORTANT! Although HX Stomp now has two additional block *locations, this doesn't mean it magically has more DSP horsepower to accommodate those blocks.* Don't be surprised if you're not able to fill all eight blocks with exactly what you want, especially considering that the polyphonic pitch models take up a lot more DSP. Also, it shouldn't come as a surprise, but *any presets you make in 3.0 will not be compatible with 2.92 or earlier firmware.*

Command Center in HX Stomp

(HX Stomp; feature already in Helix Floor, Helix Rack/Control, Helix LT, and HX Effects)

Why would people expect a stompbox with only three footswitches to also act as a master MIDI controller, but it has its uses. For example, you can use HX Stomp as both an audio interface and a remote control for YouTube lesson videos by sending QWERTY hotkeys. To set up HX Stomp:

1. Press together to open the **Menu,** and then press **Knob 3 (Command Center).**
2. Turn the **Upper Knob** to select the command source **(Instant 1-6, Footswitch 1-5, or EXP 1-2).**
3. Turn **Knob 1 (Command)** to select the type of command you want to transmit. For information on the types of commands HX Stomp transmits, see the *HX Stomp 3.0 Owner's Manual*.

Tuner in Helix Native

(Helix Native; feature already in Helix and HX hardware)

Helix Native now has the same needle and strobe tuner as Helix. To open the tuner:

1. Click the tuner icon.
2. Choose the tuner type by selecting **Coarse, Fine,** or **Strobe.**

Gain Reduction Meters in Helix Native

(Helix Native; feature already in Helix hardware)

The following block types display a gain reduction meter in the inspector when selected:

- *Dynamics > Compressor*
- *Dynamics > Gate*

Artist Presets

IMPORTANT! Because we never want to overwrite your own presets, upgrading to 3.0 does NOT automatically make these presets visible; you will need to manually restore them. Restoring factory setlists and presets will completely overwrite your own, so make sure they're backed up first!

Tip: With Helix Floor/Rack/LT/Native, if you want to keep both your presets and the latest factory presets, back up your custom presets as setlists and after the procedure below, drag your setlists onto unused setlist locations.

After backing up your presets, turn off Helix/HX. Then, perform the following procedure:

- **Helix Floor/LT:** While holding footswitches 7 and 8 (two leftmost switches on the bottom row), turn on Helix Floor/LT. Wait for "Will restore..." to appear, and then let go.

- **Helix Rack/Control:** While holding knobs 3 and 4 (two middle knobs below the screen), turn on Helix Rack. Wait for "Will restore..." to appear, and then let go.
- **Helix Native:** Click the gear icon in the lower left corner, click the Presets/IRs tab, and click "Restore Factory Setlists."

Helix 3.0 includes factory presets created by the following artists:

Helix Floor, Helix Rack/Control, Helix LT, Helix Native FACTORY 1 setlist

13B-14C—Fish (Christina Aguilera)
14D—Jeff Waters (Annihilator)
15A, 15B—Richie Castellano (Blue Öyster Cult)
15C—Robbie Calvo
15D—Ryan "Fluff" Bruce (Dragged Under)
16A-16D—Felix Martin
17A—Mario Quintaro (Spotlights)
17B, 17C—Bumblefoot (Sons of Apollo)
17D, 18A—Billy Sheehan (Sons of Apollo, Winery Dogs)
18B—Andy Abad (Jennifer Lopez)
18C-19B—Misha Mansoor (Periphery)
19C—Duke Erikson (Garbage)
19D—Steve Marker (Garbage)
20A—Eric Avery (Garbage)
20B—Bill Kelliher (Mastodon)
20C—John Browne (Monuments)
20D—Olly Steele (Monuments)
21A—Jon Button (The Who)
21B—Trev Lukather (Levara)
21C—Steve Howe (Yes)
21D—Dustin Kensrue (Thrice)
22A—Pete Thorn
22B—Rhett Shull
22C—Jade Puget (A.F.I.)
22D, 23A—Jeff Schroeder (The Smashing Pumpkins)
23B—Graham Coxon (Blur)
23C, 23D—Chris Buck
24A—Lewis Allen (Sam Smith)
24B-25A—Rabea Massaad
25B—Soren Andersen
25C—Devin Townsend
25D—Nathan Navarro
27A—Vernon Reid
27B—Philip Bynoe
27C, 27D—Markus Reuter

HX Effects

12A-12D—Julien Baker
13A—Sarah "Noveller" Lipstate
13B, 13C—Steve Stevens

Other Changes and Improvements

- On HX Stomp, *Global Settings > Footswitches > Stomp Select* now has an "Off" value. When set this way, *neither* touching nor pressing will change which block is in focus (value is already in Helix Floor, Rack, and LT).
- Favorites can now be restored from a backup, independently of setlists, presets, IRs, and Global Settings.
- User Model Defaults can now be restored from a backup, independently of setlists, presets, IRs, and Global Settings.
- Gain reduction meters now also appear for the Input block's noise gate.
- Gain reduction meters and signal present indicators are more responsive.
- Parameter knob ballistics have been improved.
- Parameter slider handles are thinner to accommodate new min/max value indicators.
- Press-turning a knob to assign Snapshot control now requires a bit more turning, to minimize accidental assignments.
- *Pitch/Synth > 4 OSC Generator* OscFreq parameters now display single Hz resolution between 100 and 1000 Hz.

3.10 Update

Increased Oversampling Throughout

(Helix Floor, Helix Rack/Control, Helix LT, Helix Native, HX Effects, HX Stomp, HX Stomp XL)

Oversampling has been increased across the board, resulting in higher fidelity, fewer aliasing artifacts, and smoother decay trails, especially when running multiple distortion stages or with higher gain tones. Many people may not notice a difference, but those highly sensitive to aliasing will appreciate 3.10's smoother response. Amp and effects models have been optimized to accommodate these improvements without increasing DSP usage.

Customizable Stomp Switches

(HX Stomp, HX Stomp XL; feature already in Helix Floor, Helix Rack/Control, Helix LT, and HX Effects)

As long as something is assigned to a Stomp switch (bypass one or more blocks, toggle parameter Min/Max, or Command Center command), switches in Stomp Footswitch Mode can now have custom labels and colors. This is especially helpful when multiple blocks, controls, or commands are assigned to the same switch. To assign labels and colors:

1. Touch a stomp mode footswitch to select it. From either the **Bypass Assign** or **Command Center** menu, press **PAGE>** and then **Customize.**
2. Use the **Upper Knob** and **Knob 2 (Character)** to name the stomp.
3. Press **PAGE>** and turn **Knob 3 (Switch LED)** to choose a custom color. When set to "Auto" (the default), the switch reflects its assignment (Delays are green, Filters are purple, Commands are white, etc.)
4. When finished, press **<PAGE** and **Knob 3 (OK).**

Customizable Snapshot Switches

(HX Stomp, HX Stomp XL; feature already in Helix Floor, Helix Rack/Control, Helix LT, and HX Effects)

Switches in Snapshot Footswitch Mode can now have custom labels and colors. To assign labels and colors:

1. From **Play View,** press the **Upper Knob** to open the **Preset List.**
2. Turn **Knob 2 (Snapshot)** to select the snapshot you want to rename, and press **ACTION.**
3. Press **Rename Snapshot.**
4. Use the **Upper Knob** and **Knob 2 (Character)** to name the stomp.
5. Press **PAGE>** and turn **Knob 3 (Switch LED)** to choose a custom color. When set to "Auto" (the default), the switch is white.
6. When finished, press **<PAGE** and **Knob 3 (OK).**

Tuner Trails

(Helix Floor, Helix Rack/Control, Helix LT, HX Effects, HX Stomp, HX Stomp XL)

Don't sleep on this one: The Tuner screen has a new "Trails" parameter. When set to "On," delay repeats and reverbs' decay continue to ring out when the tuner is engaged (even the Looper keeps running). No more awkward "everyone's staring at me because my B string went wonky."

Expanded 3.0 Models

(Helix Floor, Helix Rack/Control, Helix LT, Helix Native, HX Effects, HX Stomp, HX Stomp XL)

3.0 added a boatload of new creative effects, and 3.10 adds new parameters to some of them to expand their flexibility and fun factor. These parameters are already included in this eBook's descriptions.

However, these parameters will not show up in presets created before version 3.10. They'll appear only if you remove the block (or change the model and reload it). Also, if you've saved any of the following models' defaults, they will need to be re-tweaked and saved again:

- Delay > **Poly Sustain**
- Delay > **Glitch Delay**
- Pitch/Synth > **12 String**

Instant Command Wait Time

(Helix Floor, Helix Rack/Control, Helix LT, HX Effects, HX Stomp, HX Stomp XL)

On the Command Center page, Instant commands (lightning bolt icon) have a new "Wait Time" parameter. This lets you delay certain events by up to 1000 ms. This can help Helix/HX control older MIDI gear that may behave incorrectly when receiving multiple commands that are spaced very close together. Unless you specifically need to delay a message, leave this set to "0 ms."

Improved MIDI Implementation

(Helix Floor, Helix Rack/Control, Helix LT, HX Effects, HX Stomp, HX Stomp XL)

These improvements include:

- MIDI Snapshot changes on CC69 that are received during preset loads will now be buffered and executed once the preset load is finished. So, you can send a MIDI Snapshot change immediately after a Program Change message. This allows loading a preset with a different Snapshot than the one with which it was saved.
- Bank/Preset/Snapshot Up and Down messages how properly work via incoming FS1/FS7 emulation MIDI messages.

- FS1/2/3 emulation MIDI messages can now be momentary for HX Stomp (values 64-127 = press; values 0-63 = release).
- Stomp switch emulation MIDI messages now work while in preset or snapshot footswitch mode.
- New MIDI message (Helix Floor/Rack/LT/HX Stomp XL): CC71 engages the MODE switch.
- New MIDI message: CC 72 value 64-127 = next preset, value 0-63 = previous preset.
- New MIDI message (Helix LT, HX Stomp, HX Stomp XL): CC 73 toggles between Play and Edit views.

Other Changes and Improvements in 3.10

- HX Edit and Helix Native now add support for macOS 11 Big Sur.
- Helix Floor and Rack/Control only: The scribble strip above FS6 (MODE) now displays which mode is currently active—Preset or Stomp. You can still hold FS6 to engage hands-free Pedal Edit mode.
- On the Controller Assign page, turning Knob 2 (Controller) no longer resets the Min Value and Max Value.

3.15 Update

New Legacy Effects in 3.15

(Helix Floor, Helix Rack, Helix LT, Helix Native, HX Effects, HX Stomp, HX Stomp XL)

18 additional effects—most from the FX Junkie model pack for POD Farm 2.5—have been added to the Legacy subcategory in their respective effects categories. Don't sleep on these!

Distortion > **Bronze Master** (Legacy), based on the Maestro® Bass Brassmaster. Originally designed for bass, but equally cool on guitar, the Maestro® Bass Brassmaster is considered by many to be the Holy Grail of bass distortion units, an ultra-rare bird designed in the early 70's for Maestro® by synth genius Tom Oberheim. **Note:** The Blend parameter is not like overall distortion Mix. Instead, it sets how much of the filtered signal passes through the clipping/octave circuitry.

Distortion > **Killer Z** (Legacy), based on the BOSS® Metal Zone MT-2. Equipped with a dual gain circuit, the MT-2 provides amazing sustain plus heavy mids and lows similar to a stack of overdriven amps. We've simplified the EQ controls a bit to make the Killer Z model, but you'll still find the sought after flavor of the MT-2 style sound.

Modulation > **Tape Eater** (Legacy), Line 6 Original. If you've ever had a cassette player eat a tape before you'll know what we're talking about. Try this with a slow speed setting and a 100% wet mix.

Modulation > **Warble-Matic** (Legacy), Line 6 Original. This effect is reminiscent of the Sweeper model, but when used subtly it can produce a nice mild phasey sound or with Depth maxed out you can simulate the sound of an alien spacecraft landing in one of those old 50's sci-fi movies.

Modulation > **Random S&H** (Legacy), Line 6 Original. This has a similar effect as the old Oberheim® Voltage Controlled Filter. It creates changes in tone by randomly emphasizing certain frequencies. Try pressing the Speed knob to lock it to tempo and playing single chords to that tempo.

Modulation > **Sweeper** (Legacy), Line 6 Original. Imagine having two wah pedals on steroids separated in a stereo field that are pulsating in opposite positions and you're close to what you'll hear here. Use the Q and

Freq parameters to set the character of the sweep and adjust Depth to go from subtle to full on freak out. Any resemblance to guitar tracks heard in a particular genre of B films is strictly coincidental.

Delay > **Phaze Eko** (Legacy), Line 6 Original. Starting with the basic tone of our EP-1 tape delay emulation, they've added something very much like a Uni-Vibe to the delay repeats. The result is an echo unit that gives you unique new creative possibilities for adjusting the tone of your delays with a beautiful, burbling texture.

Delay > **Bubble Echo** (Legacy), Line 6 Original. Bubble Echo has a sample-and-hold filter on the repeats. It takes a filter sweep (like the one on Sweep Echo), chops it up into little bits, and rearranges them semi-randomly, so that it sounds like sudden little bits of wah pedal randomly sprinkled about.

Pitch/Synth > **Synth Lead** (Legacy), Line 6 Original. These are styled after popular analog monophonic synth lead sounds from Moog, ARP and Sequential Circuits.

Pitch/Synth > **String Theory** (Legacy), Line 6 Original. This emulates classic synth string sounds like those found in the ARP Solina String Ensemble and the Elka® Synthex. The harder you pick, the brighter the sound. We somehow had two separate effects called "Synth String"—one from POD Farm 2.5 and the other from FM4, which was already added to Helix/HX in 1.50. Renamed the POD Farm version "String Theory" to avoid confusion.

Pitch/Synth > **Synth FX** (Legacy), Line 6 Original. These sounds aren't really designed to be musical. These are more "special effects" sounds. You'll hear a lot of these kinds of sounds in movie soundtracks.

Pitch/Synth > **Buzz Wave** (Legacy), Line 6 Original. These are cool combinations of saw and square waves with fast vibrato. The 8 different Wave parameters offer different vibrato speeds and different pitches.

Pitch/Synth > **Rez Synth** (Legacy), Line 6 Original. These are all sweeping low pass filter effects with the resonance set high. Resonance is a peak at the frequency of the low pass filter.

Pitch/Synth > **Saturn 5 Ring Mod** (Legacy), Line 6 Original. Ring modulators take two signals (one supplied by your guitar, the other supplied by the effect) then adds and subtracts their frequencies. Electro-Harmonix® makes a ring modulator pedal called the Frequency Analyzer that is a popular guitar effect. The only limiting factor is that the pitch of the signal provided by the effect is constant. So, you have to play only in the key incorporating that pitch to be musical.

Pitch/Synth > **Double Bass** (Legacy), Line 6 Original. This effect has two oscillators that track the pitch of your guitar—one square wave tuned one octave down, and one saw tooth wave two octaves down.

Pitch/Synth > **Seismik Synth** (Legacy), Line 6 Original. This effect has an oscillator that tracks the pitch of your guitar. You can choose between 8 different wave shapes which give you different "flavors"—all of them one or two octaves down from the original pitch.

Pitch/Synth > **Analog Synth** (Legacy), Line 6 Original. These are great for funky synth guitar (or bass) lines. These sounds were made popular by Moog and ARP.

Pitch/Synth > **Synth Harmony** (Legacy), Line 6 Original. If you loved those big synth leads from 70's era prog bands, then you'll love this effect. There are two synth waves at work here. Your first two parameters allow you to choose a pitch interval of your original note played. The Wave parameter works differently from what you'd expect with the other synth models; here it controls the gain of the saw wave, while the square wave gain remains constant.

Input Block > Per-Preset Guitar Pad

(Helix Floor, Helix Rack/Control, Helix LT, HX Stomp, HX Stomp XL)

The *Input > Multi* and *Input > Guitar* block now has an additional Guitar Pad parameter that lets you engage the analog guitar pad per preset ("Off" or "On). Or you could leave it set to "Global" and continue to turn the guitar pad on and off from the Global Settings menu. **Note:** For HX Stomp and HX Stomp XL, "Global" refers to the *Global Settings > Ins/Outs > Input Level* setting. When Pad is on, it's the same as if Input Level is set to "Line.". When off, it's the same as if Input Level is set to "Inst[rument]."

Global MIDI Control of Knobs 1-6, <PAGE/PAGE>, and Preset Save

(Helix Floor, Helix Rack, Helix LT, HX Effects, HX Stomp, HX Stomp XL)

Instead of having to assign parameters manually to incoming MIDI CC messages, **Knobs 1-6** for any block on the Home screen (**Knobs 1-3** on HX Effects, HX Stomp, or HX Stomp XL) respond to new global MIDI CCs. This turns any programmable MIDI keyboard or tabletop controller into a simple way to edit blocks without having to reach down (for those of you who can't be bothered to use Pedal Edit Mode). In addition, sending Helix/HX a CC74 message will save any changes to the preset. ***IMPORTANT!*** We ran out of reserved Global CCs, and unfortunately had to steal five user-assignable CCs for this feature. Any parameters you've assigned to CC77-CC81 will have to be remapped. Sorry 'bout that.

Other Changes and Improvements in 3.15

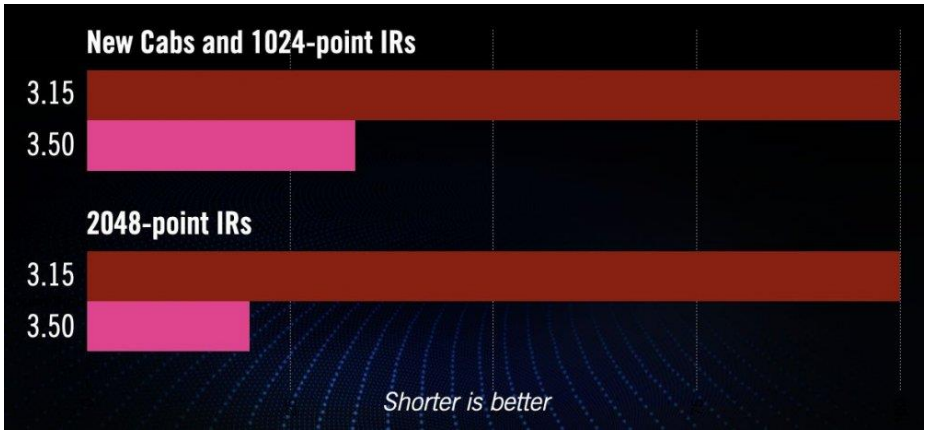
- **HX Stomp XL** only: *Global Settings > Footswitches > Snapshot Mode* adds a third value—Toggle, like the one in HX Effects and POD Go. This remembers whether Preset Mode is in Preset or Snapshot mode and allows for toggling between Stomp and Snapshot mode via the MODE switch.
- The max delay time for several *Delay > Legacy* models has been increased from 2 seconds to 2.5 seconds.
- Numerous additional minor improvements to *Delay > Legacy* models
- *Delay > Multitap 6* has been optimized to use less than half the DSP compared to pre-3.15 builds.
- MIDI CC69 values are now transmitted upon snapshot change.
- *Command Center > HX Snapshot* commands are now affected themselves by snapshot changes. This means the same Stomp switch could recall a different, specific snapshot (or next or previous snapshot) *per snapshot.*
- *Command Center > HX Snapshot* command parameters can now be controlled by Snapshots themselves. For example, suppose you are on Snapshot 1. and create an HX Snapshot command on FS2. You set it so that pressing that switch recalls Snapshot 3. You then switch to Snapshot 3 and set the same switch to recall Snapshot 6. Recall Snapshot 6 and set the switch to recall Snapshot 2. Finally, you recall Snapshot 2 and set the switch to recall Snapshot 1. Now, you go back to Snapshot 1 and begin pressing the switch. Each successive press recalls the assigned Snapshot, and you would cycle Snapshot 1 > 3 > 6 > 2 > 1 > 3 > 6 > 2 and so on for each switch press. **Note:** This assumes that the Snapshot Edits global is set to Recall. If it's set to Discard, you'd have to save the preset after each set for the changes to stick.
- *Bypass Assign > EXP Pedal 1/2* now has a new Behavior parameter to control how bypassing is handled. The default value "Toggle" behaves as this feature always has—bringing the pedal past threshold will toggle the block's bypass state from what it is currently (i.e., enable if currently bypassed, and vice versa). "Toe Down" and "Heel Down" always bypass the block at the designated position, regardless of the block's starting state. So, with "Heel Down" selected, the block will always enable when you cross the Position threshold, and bypass when you return below it.

- We ran out of reserved Global CCs. and unfortunately had to steal five user-assignable CCs for MIDI control of **Knobs 1-6** and **<PAGE/PAGE>**. Any parameters you've assigned to CC77-CC81 will have to be remapped.
- Encoder ballistics have been improved. For parameters with hundreds of values, you can now go from Min to Max in a couple of turns.

3.50 and 3.60 Updates

New Cabs and IRs now use 66-80% Less DSP

(Helix Floor, Helix Rack/Control, Helix LT, Helix Native, HX Effects [IRs only], HX Stomp, HX Stomp XL)



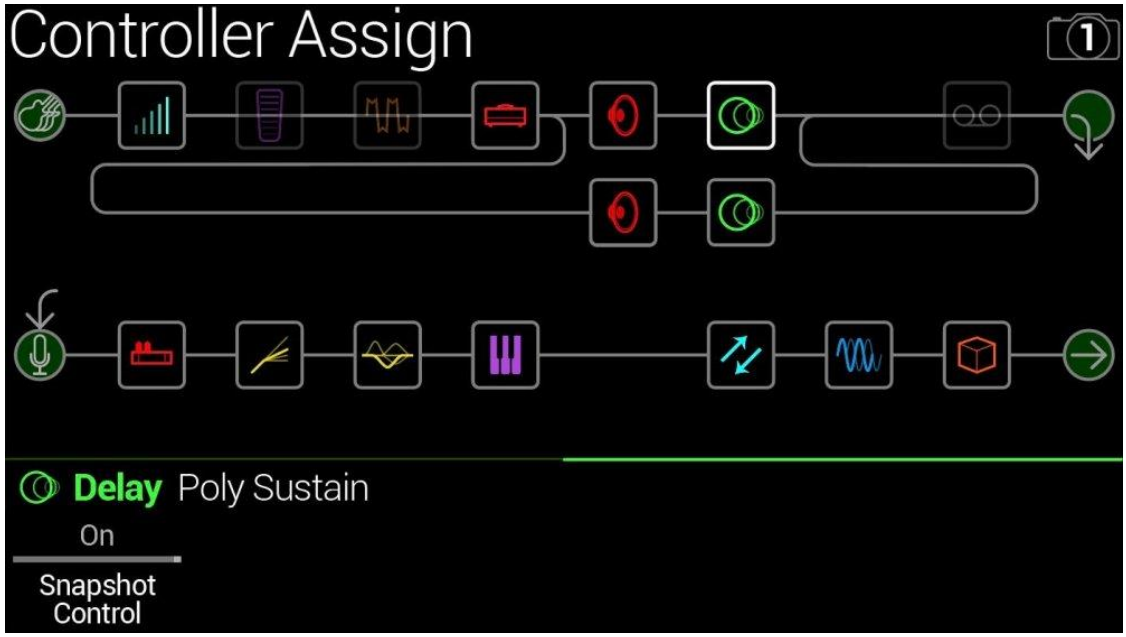
Due to further improvements to Helix Core, new Cabs and 1024-point IRs use roughly 66% less DSP than version 3.15 or earlier. Even the new IR > Dual block uses less DSP than a single IR block in 3.15. 2048-point IRs use roughly 80% less DSP, although they still use a lot of memory, so they're still limited to one instance per path. **Note:** Older Hybrid cabs use the same amount of DSP as in earlier firmware. This change affects only the cabs added in 3.50.

Disable Snapshot Control over Parameter Assignments

(Helix Floor, Helix Rack/Control, Helix LT, HX Effects, HX Stomp, HX Stomp XL)

Prior to 3.50, when assigning a parameter to a physical control (EXP 1, footswitch, etc.) or MIDI, Snapshots are also always assigned. In 3.50, they still are, but there's an additional "Snapshot Control" parameter on the Controller Assign page that lets you bypass this behavior:

1. From the **Controller Assign** menu, select the desired parameter and turn **Knob 2 (Controller)** to select any value other than **"None"** or **"Snapshots."** A second page appears.
2. Press **PAGE>** and turn **Knob 1 (Snapshot Control)** to **"Off."** It's set to "On" by default for all parameters.



Other Changes and Improvements in 3.50

- Encoder ballistics have been drastically improved. For example, tone stack values can go from 0.0 to 10.0 with one turn.
- ***Helix Floor/Rack/LT only:*** New Shortcut. While the Model List is open, pressing [AMP] jumps to highlight the Amp category without having to scroll down to it.
- ***Helix Floor/Rack/LT only*:** The Impulse Response category's name has been shortened to "IR" (which now matches that of HX Stomp/XL). This accommodates a wider subcategory column in the model list and allows for longer IR names in the inspector header.

Other Changes and Improvements in 3.60

- Because 3.60 adds a second Grammatico amp, the three existing Grammatico LaGrange amp models have been renamed to "GrammaticoLG Nrm," "GrammaticoLG Brt," and "GrammaticoLG Jmp." Note that presets included in v1.2.1 that included the Grammatico amps have been renamed to reflect the new names.
- A few 3.50 factory presets have been updated to utilize 3.60 cabs where appropriate.

3.70 Updates

Cab Delay > Auto Value

(Helix Floor, Helix Rack/Control, Helix LT, HX Stomp, HX Stomp XL)

Helix/HX's new cab engine (introduced with 3.50) makes it easy to quickly find a great cab sound by automatically phase and time-aligning the mics in dual cab setups; this avoids the phasing or "thinning out" that can occur naturally when using two mics at different distances. However, some studio engineers are very adept at using phase misalignment to achieve interesting sounds. We've now added a new "Auto" value to the Cab > Delay parameter (all the way left, before 0.0 ms) that automatically approximates the delay through the air when changing the mic Distance parameter.

For most people, however, we recommend leaving Delay set to the default "0.0 ms."

Known Issues in 3.70

- The mono version of Reverb > Nonlinear exhibits a less-smooth decay than the stereo version, which is especially noticeable on percussive material. It also affects the stereo version if merged to a mono path or when utilizing a mono output.
- In some cases if a Path 1B or 2B Input block is set to the same return as a block on a B path in a corresponding preset, the preset can load blank.
- After performing a Factory Reset, attempting to load 128 or more IRs can result in a "Failed to get impulse names" -8207 error. Instead, load fewer than 128 IRs at once.
- If Looper > 1 Switch Looper is playing, the first MIDI CC62 (Play Once) command is ignored. Sending a second CC62 command will work properly.
- Helix Native (Windows VST/AAX) only: Switching from certain reverb types to others (for example, from Hot Springs to Shimmer or Glitz to Shimmer) can result in DSP loss.
- Helix LT only: After entering and exiting the tuner from Snapshot performance view, Helix LT returns to the home screen. In the meantime, note that pressing FS6 (MODE) and FS12 (TAP/TUNER) together returns to Snapshot performance view.
- HX Stomp/HX Stomp XL only: If you have a snapshot command set to a footswitch (and Global Settings > Displays > LED Rings is set to "Off/Brt"), the footswitches may not light up red when pressed.
- HX One only: User Model Defaults created with 3.60 firmware are lost after update to 3.70. IMPORTANT! Be sure to use HX One Librarian to back up all data BEFORE updating to 3.60!
- HX One only: In rare cases, very quickly push-turning the EFFECT encoder can maintain the effect category carousel view. Press HOME to exit.
- HX One only: When using HX One in 4-Cable Method, stereo models can collapse to mono when the Insert position is set to Post.
- HX One only: When True Bypass is engaged, the Tuner does not operate correctly.

3.80 Update

New HX One Features in 3.80

- HX One only: The On/Off footswitch state is now saved with the preset
- HX One only: You can now view the preset name instead of the model name in the Home view. Configure this in the new Global > Display setting:
    - Auto (default): Displays the preset name by default. When scrolling through models using the effect encoder, the model name appears and persists for 3 seconds after the last adjustment before reverting to the preset name. When editing model parameters, the preset name briefly switches to the model name, persisting for 3 seconds after the last edit before returning to the preset name
    - Model: Always displays the model name
    - Preset: Always displays the preset name.
- HX One only: Added support for MIDI control of the Shuffling Looper
- HX One only: Introduced a new MIDI command for preset navigation (CC72):
    - CC72, value 0-63: Previous preset
    - CC72, value 64-127: Next preset
- HX One only: The chromatic tuner is now available when HX One is set to True Bypass

Bug Fixes in 3.80

- Delay > Cosmos Echo would apply the same wow/flutter modulation to the left and right delays, making it appear in certain cases as though the Spread control was inoperable.
- If you let your Helix/HX box sit, doing nothing for many hours, certain delay and modulation effects could stop processing audio until the preset was reloaded.
- Delay > Poly Sustain could sometimes exhibit glitching audio if Random or Random Speed knobs were maxed out.
- If the Amp > Ampeg B-15NF's Bass parameter was set to 10.0, the model could gradually lose volume over the course of 30 minutes or so.
- If the Amp > Mandarin Bass 200's Bass parameter was set to 0.0, the model could lose volume over the course of 30 minutes or so.
- HX Stomp/XL only: In certain configurations, output USB 5/6 would not always mirror the instrument input.
- HX One only: There was an issue with Looper > Shuffling Looper when exiting the preset list by pressing both footswitches.
- HX One only: Stereo models could collapse to mono when the insert position was set to Post—*FIXED*
- HX One only: Looper undo/redo states could become reversed when controlled via MIDI—*FIXED*
- HX One only: Certain Delay effects could result in glitches when receiving MIDI Clock changes while Note sync is disabled.
- HX One only: In some cases, Flux Min and Max values may not update after resetting the parameter to its default.
- Other minor fixes and improvements

Known Issues

There is a known issue in HX Edit 3.80 on Apple silicon computers (M1, M2, M3, M4) where some global settings are not properly restored from a backup file. A hotfix for this issue is forthcoming. Windows and Intel Apple computers are not affected by this issue.

Additional Resources

There are many Helix-related online resources. These are some of my favorites.

Line 6 Community
This is part of Line 6's support system. It's an excellent resource for peer-to-peer support. Both newbies and long-time Line 6 product users with deep, and useful, knowledge frequent the community.

Line 6 Helix Resources
You'll find official videos from Line 6, blog posts, and a link to a comparison chart for Helix processors.

Helix Help
Unofficial guide for the Line 6 family of Helix products with model lists and descriptions, tips, reference guides, and release notes.

Helix Discussion (vguitarforums.com)
This forum is a wealth of knowledge about electronic and high-tech guitar. It's independent, with no corporate affiliation, and is run by volunteers. Although not specific to Helix, it's a repository of interesting information for guitar players who want to go beyond conventional guitar.

Browse Helix Models and Effects
The Fluid Solo web site lists and describes the various amp models and effects in Helix. It includes images of the amps that are modeled, along with descriptions and the names Helix gives them.

Legacy Effect Descriptions
The linked PDF includes information on many of the Legacy effects included in Helix.

Legacy Effect I/O Handling
John Bercik (coachz on the Helix forum analyzed all the Legacy effects regarding how they respond to mono and stereo inputs. I've adapted his post (linked to above) to an Excel spreadsheet on the next page, and reproduced the information with his permission. Here's what the abbreviations stand for:

- M > M = Mono in, Mono out
- M > S = Mono in, Stereo out
- S > S = Stereo in, Stereo out
- S > M = Stereo in, Mono out

Thanks, John!

Effect	Mono In	Stereo In
All Distortions	M > M	S > M
All Dynamics	M > M	S > M
All Pitch/Synth	M > M	S > M
All Reverbs	M > S	S > S
Modulation		
Pattern Tremolo	M > S	S > S
Panner	M > S	S > S
Bias Tremolo	M > S	S > S
Opto Tremolo	M > M	S > S
Script Phase	M > M	S > S
Panned Phaser	M > S	S > S
Barberpole	M > M	S > S
Dual Phaser	M > S	S > S
U-Vibe	M > M	S > M
Phaser	M > S	S > S
Pitch Vibrato	M > M	S > M
Dimension	M > S	S > S
Analog Chorus	M > S	S > S
Tri Chorus	M > S	S > S
Analog Flanger	M > S	S > S
Jet Flanger	M > S	S > S
AC Flanger	M > M	S > S
80A Flanger	M > M	S > M
Frequency Shift	M > M	S > S
Ring Modulator	M > S	S > S
Rotary Drum	M > S	S > S
Rotary Drm/Horn	M > S	S > S
Tape Eater	M > M	S > M
Warble Matic	M > M	S > M
Random S&H	M > M	S > M
Sweeper	M > S	S > S

	Mono In	Stereo In
Delay		
Ping Pong	M > S	S > S
Dynamic	M > M	S > S
Stereo	M > S	S > S
Digital	M > M	S > S
Dig w/ Mod	M > M	S > S
Reverse	M > M	S > M
Lo Res	M > M	S > S
Tube Echo	M > S	S > S
Tape Echo	M > S	S > S
Sweep Echo	M > M	S > S
Echo Platter	M > S	S > S
Analog Echo	M > M	S > S
Analog w/ Mod	M > S	S > S
Auto-Volume Echo	M > S	S > S
Multi-Head	M > M	S > S
Bubble Echo	M > M	S > M
Phaze Eko	M > M	S > M
Filter		
Voice Box	M > M	S > M
V Tron	M > M	S > M
Q Filter	M > M	S > S
Seeker	M > M	S > S
Obi Wah	M > S	S > S
Tron Up	M > M	S > S
Tron Down	M > M	S > S
Throbber	M > M	S > S
Slow Filter	M > M	S > S
Spin Cycle	M > S	S > S
Comet Trails	M > S	S > S

About the Author

Musician/author Craig Anderton is an internationally recognized authority on music and technology. His onstage career spans from the sixties with the group Mandrake, through the 2000s with electronic groups Air Liquide and Rei$$dorf Force, to the "power duo" EV2 with Public Enemy's Brian Hardgroove, and EDM-oriented solo performances.

Craig has played on, produced, or mastered over twenty major label recordings, did pop music session work in New York on guitar and keyboards, played Carnegie Hall, and more recently, has mastered well over a hundred tracks for various artists in addition to producing, mixing, and mastering his own projects. He also brings his production skills to soundtracks for a variety of video projects.

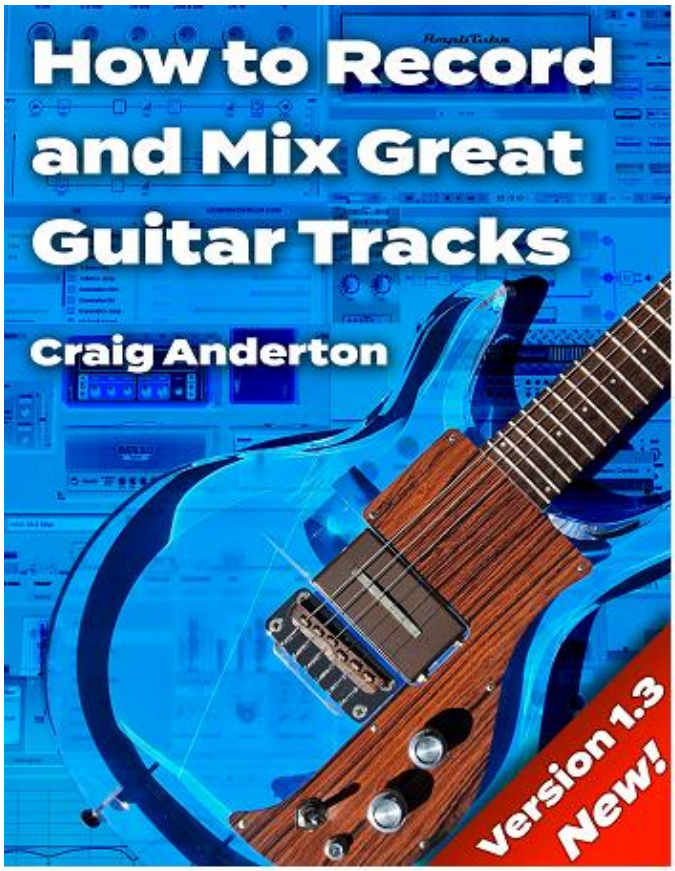
In the mid-eighties, Craig co-founded *Electronic Musician* magazine. As an author, he's written over fifty books on music technology, and over a thousand articles for many popular websites as well as magazines like *Sound on Sound, Rolling Stone, Pro Sound News, Guitar Player, Mix,* and several European publications.

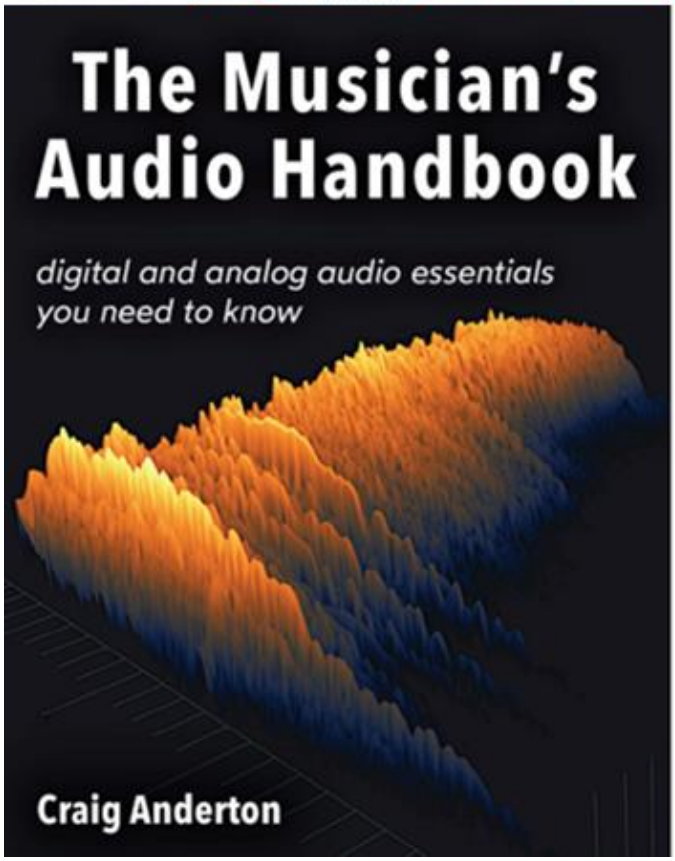
Craig has lectured on technology and the arts (in ten countries, 38 U.S. states, and three languages), and done sound design work for many companies including Alesis, Gibson, Peavey, PreSonus, Roland, and Steinberg. He also served as President of **The MIDI Association**, an international trade organization for manufacturers of MIDI-related products and technologies, which also features a public-facing web destination for those who work, play, and create with MIDI.

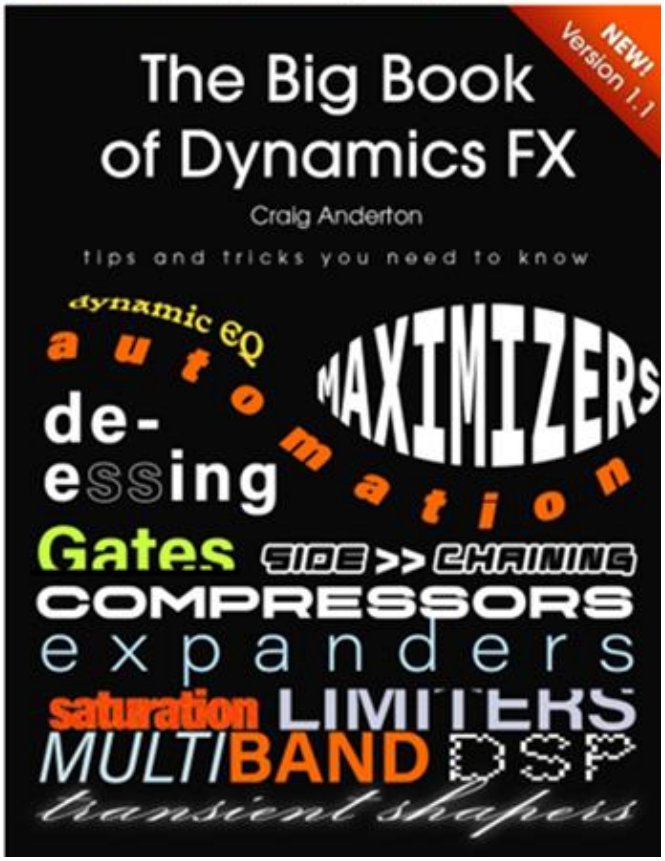
Please visit Craig's educational website at **craiganderton.org** (which also has free downloads), and listen to some of his music at **youtube.com/thecraiganderton** as well as watch some educational videos. From time to time he pops in at **instagram.com/craig.anderton**, **x.com/craig_anderton**, and **facebook.com/craig.anderton3**, but spends most of his social media time at his **Sound, Studio, and Stage** forum (now in its 30th year of continuous operation) at **musicplayer.com**. His forum also hosts the official **discussion/support forum** for Sweetwater Publishing.

Other Sweetwater Publishing Books by the Author

For more information on these best-selling books, please visit Sweetwater Publishing.







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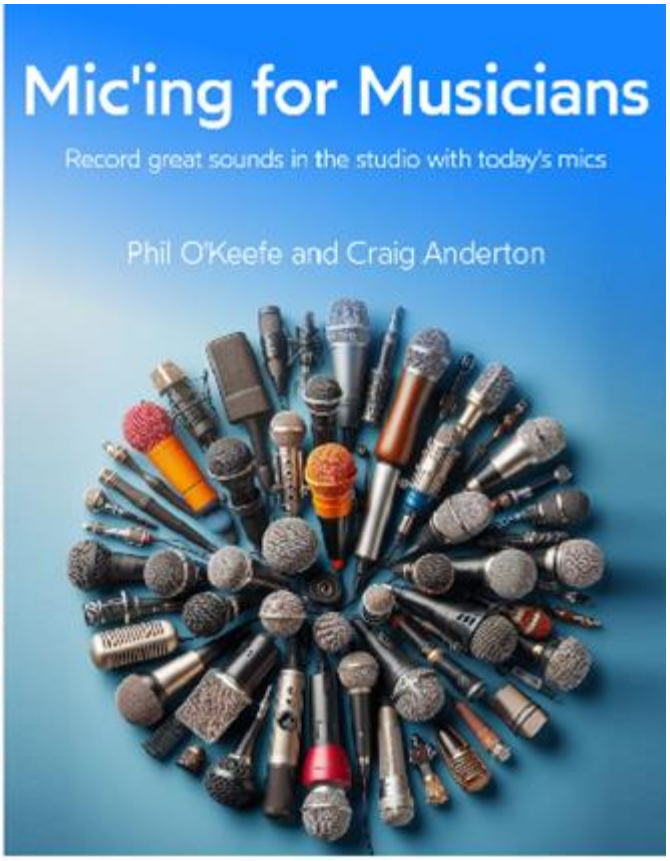
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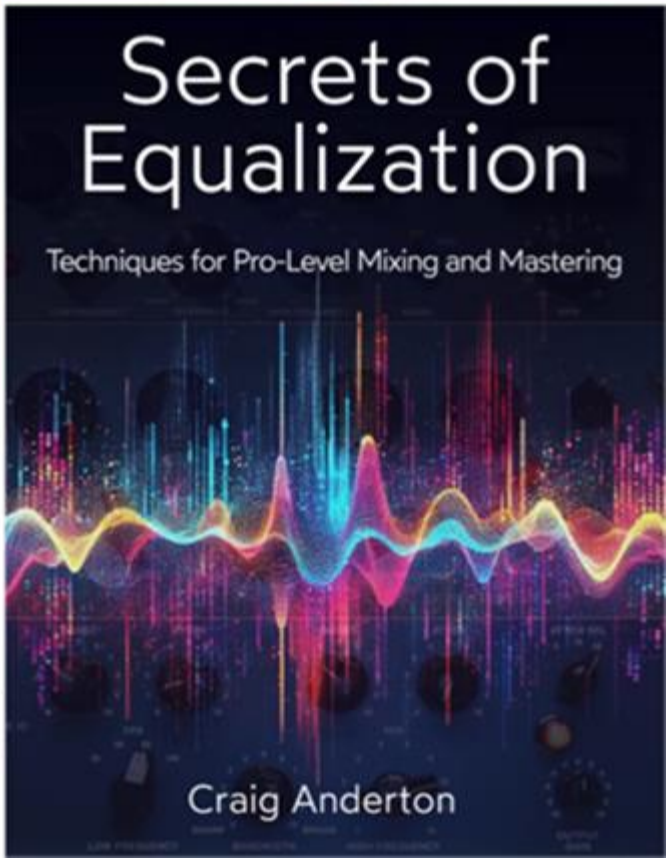
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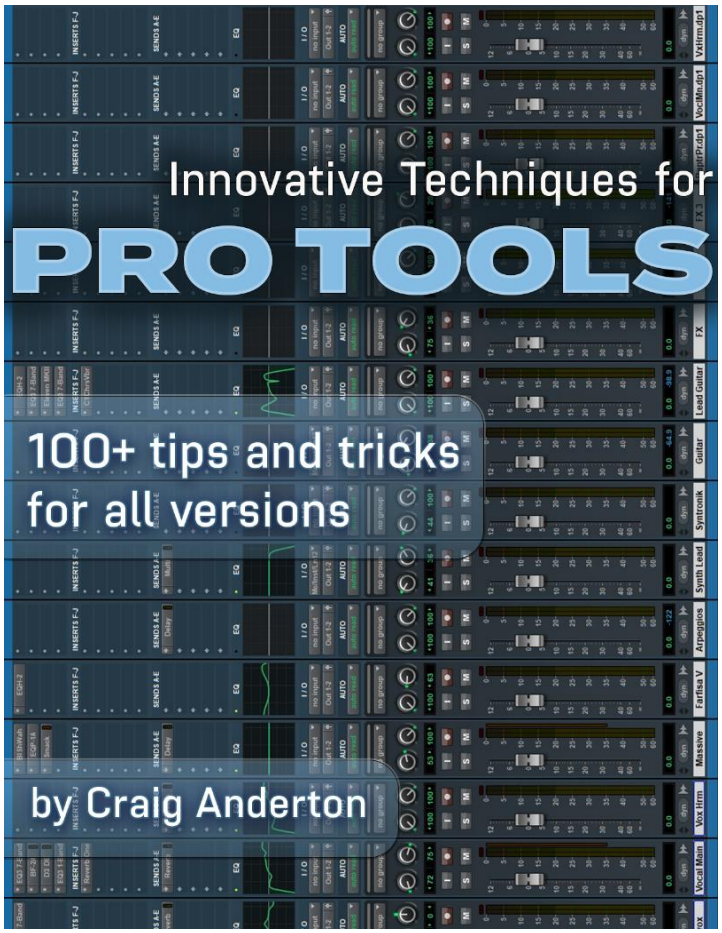
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